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Contents

VOL 9 PART 2 JULY 1990

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All-Digital Trunk Network

As we move from the 1980s to the 1990s, the benefits resulting from BT's network modernisation programme are apparent in one way or another to all who use the network. National call failures due to the network have fallen from more than 4% to less than 1% over 5 years, noisy circuits have become a thing of the past, and new services from ISDN to star services and itemised billing have been launched.

June 1990 will be recalled as a significant date in the modernisation programme having seen two milestones achieved. The first, connection of the 10 millionth customer served by a digital exchange, occurred towards the end of June, just over two and a half years from when the 1 millionth customer was connected. The second milestone, to complete the transfer of trunk traffic to the digital trunk network, was completed at the end of June. BT has become the first major telecommunications operator in the world to achieve an all-digital long-distance network. The transfer programme which took more than 5 years to complete and involved the closure of 403 outgoing analogue group switching centres is a major achievement which has combined the key competencies of network planning, advanced technology and project management.

The plans which culminated in the successful conclusion of the programme were formulated almost 20 years ago when a steering group, set up to investigate the benefits of modernisation, recommended that digital switching and transmission systems should be employed in the trunk network of the future. The recommendation led to transmission and switched network studies being undertaken which laid down the rules for network synchronisation, signalling, transmission, and routing, and an implementation strategy for the modern trunk network.

The article on p. 82 provides a brief overview of the complexities of the transfer and how it was managed. The fact that the programme was completed on time and to a very demanding and tight schedule is a credit to the tremendous team effort of all concerned.

The programme demonstrates British Telecom's continued commitment to provide the quality of service that customers require—essential for success in a competitive environment—an issue that Iain Vallance, Chairman of British Telecom, addressed in his recent lecture at the Institution of Electrical Engineers. An article based on his lecture 'The Competitive Challenge in World Communications Markets' is given on p. 84.

C. M. EARNSHAW
Director Network, British Telecom UK

Loading the Digital Trunk Network

A. J. HART†

The transfer of trunk traffic to the digital trunk network was completed at the end of June this year. This article describes the background to this important milestone in BT's modernisation programme.

INTRODUCTION

On 28 June 1990, at 13.00 hours, the loading of the digital trunk network was completed with the transfer of BT's customers in Thurso to a remote concentrator (RCU) and thus the analogue trunk switched network consisting of 460 group switching centres and transit units was finally closed. The onload programme which commenced with the opening of Coventry Spires digital main switching unit (DMSU) in March 1985 was completed within 3 months of the original target date.

BACKGROUND

BT's policy to establish a digital trunk network owes its origins to work carried out by the UK Trunk Task Force, a multi-disciplined group set up in 1967 that made extensive use of computer modelling to study trunk network development. The UKTTF established the benefits that would be derived from a digital trunk network (see Figure 1). The UKTTF also demonstrated the economic benefits of a digital network having far fewer switching nodes than the analogue trunk network. In 1971, the final report of the UKTFF was accepted by the UKTTF Steering Group, and a recommendation was made that the trunk network should be modernised by establishing an integrated digital network.

As a result of the UKTTF report, studies were carried out in the Network Planning Department during the early-1970s on the implementation of both digital transmission and switching. Transmission studies led to the development and utilisation of digital transmission systems. Switched network studies led to plans for the synchonisation network, signalling network, transmission standards, traffic routing rules, network structure, implementation strategy etc. The development of System X began in 1975 based on the work, during the 1960s/70s, of a joint British Post Office/industry study on switch development by the Advisory Group on Systems Definition (AGSD). In 1978, a strategic objective was set to link the 30 principal (business) cities by an integrated digital network by 1986.

Further studies in the early-1980s led to the introduction of Network Master Plans, the establishment of a network strategy based on 60 trunk switching nodes and the formulation of a conversion strategy for transferring traffic from the analogue to digital network.

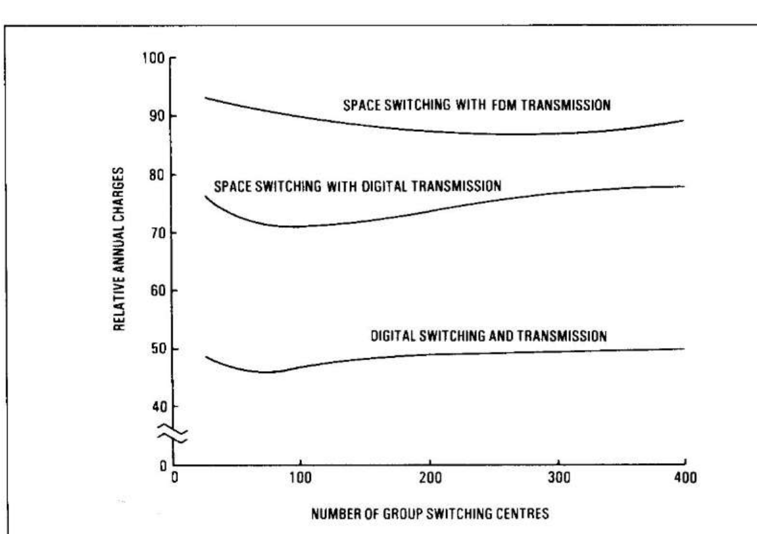
In 1983, with the creation of National Network, Trunk Services, the main thrust of trunk network modernisation began, and, in 1985, the transfer of traffic to the digital trunk network commenced under the control of the network operations centre at Oswestry.

CHALLENGE

The challenge that BT had set itself in 1985 was to transfer all its trunk traffic from analogue to digital by March 1990 and to close the analogue trunk switching units progressively as the onload advanced. It was of course necessary that the transfer should be transparent to the customer and that customers parented on either the analogue or digital trunk networks should at all times have full access. At the same time, it was important that the quality of service as perceived by the customers should improve.

This presented a complex networking task. First, the network had to be structured in such a manner as to ensure maintenance of access. As the digital trunk network was not completed until Norwich DMSU was opened in August 1987, a pipeline route had to exist between each

Figure 1
Comparative trunk-network costs



† Network Operations Centre, British Telecom

DMSU and an analogue trunk switching unit to allow calls that could not be completed on the digital network to return to the analogue network—RTA routes. Conversely, to enable calls that transversed the digital trunk network to the objective DMSU to terminate on analogue parented exchanges, DI/STD routes were provided from each DMSU to each analogue trunk switching unit. Each local unit as it was parented onto a DMSU maintained a route to its home analogue trunk switching unit so that calls originating in the analogue network could be terminated.

DIMENSIONING

Once a network structure had been installed that would allow the transition to take place in a transparent mode, it was necessary to ensure that all the routes were adequately dimensioned as the onload proceeded to ensure that the required improvements in quality of service were not put at risk. To do this, two computer support systems were developed. The online traffic information system (OTIS) monitored the traffic flows on all routes connected to the DMSUs and provided an online traffic record for the three busy hours of the day at the end of each period. The system identified exceptions against pre-set criteria so allowing early identification of potential congestion and any necessary remedial action before it impacted on quality of service. The OTIS, however, was a reactive tool; a proactive mechanism was also necessary to dimension the network in advance of loading taking place. This was a difficult task as onload was taking place daily at all the nodes in the network. To manage this problem, the advanced network modelling system OTIS II was developed. This system provided a dynamic model of the network which identified the capacity necessary on all routes connected to the DMSUs for the continuously changing loading profiles. It was the key to ensuring that sufficient distributive capacity was provided from each DMSU to terminate the traffic being loaded elsewhere in the network.

CLOSURE OF ANALOGUE TRUNK SWITCHING UNITS

There were significant benefits to be gained from closing analogue trunk switching units as the onload progressed. To achieve this, the network structure was again modified so that outgoing closure of the analogue trunk switching units could be accelerated by the provision of concentration point working, which, in simple terms, involved collectively intercepting the traffic from small analogue local exchanges before it entered the analogue trunk unit and routing it over 2 Mbit/s paths to the DMSU. To achieve the incoming closure of the analogue trunk unit, new pipeline routes were provided to the digital network known as *near-end handover routes*. When it was decided to close a particular analogue trunk unit totally, the traffic to the target unit was re-routed digitally at all other units over their near-end handover route. The work required within London for the code change was considerably simplified by rerouting the 01 code digitally by this method, thereby closing London to incoming analogue traffic prior to the changeover.

PROJECT CONTROL

In order to complete the onload programme as near as possible to the original target date, the objective was reinforced by a Board directive in the spring of 1989 to complete the programme by the end of June 1990. This was extremely challenging as, with the many competing demands for resources, the programme end date in some instances had slipped out to September 1991. Therefore, the programme had to be recovered by some 15 months within the period of one year. To achieve this, tight project control mechanisms were put in place, and through them, and significant managerial and field effort throughout the entire company, the programme has been completed by the date set by the Board.

SCALE OF THE PROGRAMME

When the programme commmenced, there were some 150 000 erlangs of traffic originating from some 6300 analogue local exchanges carried over some 400 000 circuits with 280 000 different traffic routings. It was forecast that this traffic would increase to 195 000 erlangs by the time the programme was complete, a 30% increase. In fact, by June 1990, the digital trunk network was carrying 228 000 erlangs of originated trunk traffic, a 52% increase.

CONCLUSION

The objectives set for the transfer have been achieved, it has been transparent to the customer, there has been a five-fold improvement in quality of service and the return on investment has been maximised with the progressive closure of the analogue trunk switched units. This has all been achieved against a background of unprecedented growth, and changes of strategy such as dual parenting and nodal consolidation.

Biography

Archie Hart joined BT in 1965 working first as a Level 1 in the Blackburn Telephone Area and then as a Level 2 in Leicester before moving to the network operations centre at Oswestry in late-1984. There he has been responsible for the management of the digital trunk network onload and the analogue switched trunk network closure programmes. Additionally, for 2 years he was Head of Network Management Operations involved in establishing the national network management centre and, in particular, network traffic management as a credible force in the maintenance of quality of service to the customer.

The Competitive Challenge in World Communications Markets

I. D. T. VALLANCE†

This article is based on the 7th Lord Nelson of Stafford Lecture given by Iain Vallance, Chairman of British Telecom, at the Institution of Electrical Engineers on 5 April 1990. In his address, Mr. Vallance discussed the competitive environment that characterises today's telecommunications markets, and the challenge facing British Telecom in becoming the most successful telecommunications company world-wide.

INTRODUCTION

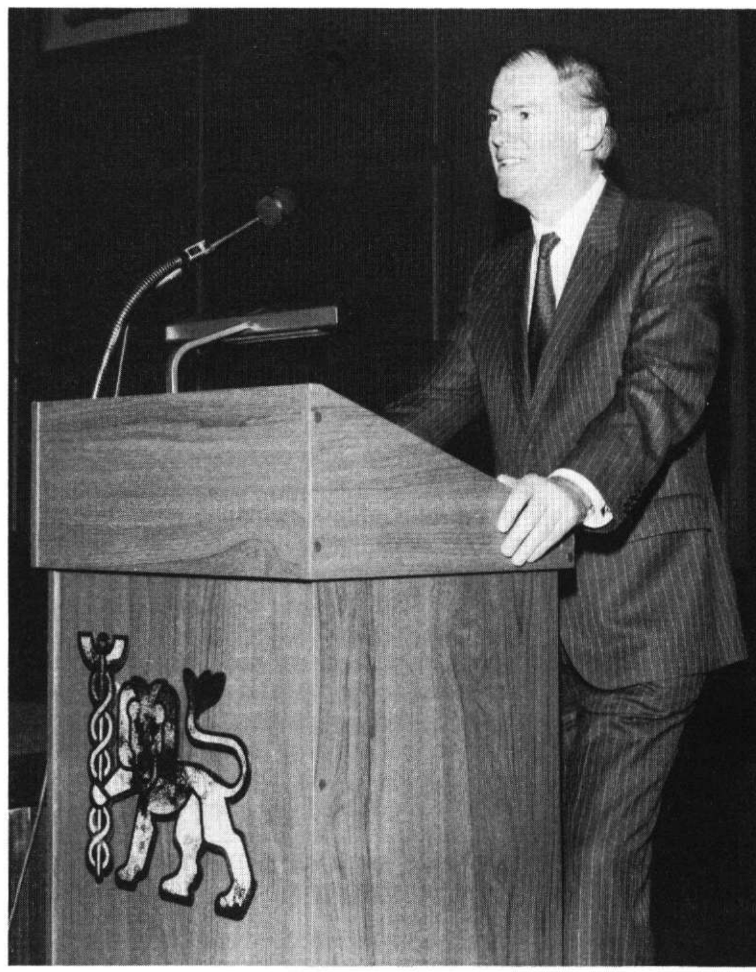
A year ago, Baroness Platt of Writtle delivered the sixth Lord Nelson of Stafford Lecture, and talked about breaking down one particular barrier: the barrier between the sexes in professional life, a barrier to opportunity both for individuals and for industry alike.

Since then all of us have seen political barriers falling at an astonishing pace. Tonight, my job is to address a challenge that still faces people as these barriers come down: the challenge we face when communications markets become truly competitive.

We are here to honour the first Lord Nelson of Stafford. A man who grasped challenges with determination. By the time I was born, in 1943, Lord Nelson had already faced up to the challenge of gearing up production for war. The defence of this country depended on the success of that effort.

He faced that challenge with an understanding of engineering that went back to his days as a student of Silvanus Thompson. But Lord Nelson's leadership, the leadership which we are celebrating, went beyond engineering because he showed the importance of paying attention to people and to the contractual arrangements that governed supplier relationships. He also faced up to the challenge of extending his leadership from the power side of electrical engineering to the light current side when English Electric took over the Marconi Wireless Telegraph Company.

I have had the privilege of working with the engineering successors of Marconi and in an office that stands on the very site of Marconi's first public transmission. British Telecom depends on electrical engineers: men, and increasingly women, both inside British Telecom, and amongst our suppliers, at home and overseas. It is not just British Telecom that depends on these engineers. Everybody who relies on telecommunications as an everyday service depends on their skill and professionalism. Their work underpins the opportunities opened up by information technology.



Iain Vallance, Chairman of British Telecom, presenting the 7th Lord Nelson of Stafford Lecture at the IEE

TELECOMMUNICATIONS OPPORTUNITIES

The opportunities they have pioneered are open to us all:

- To businessmen in the office, and on the move.

† Chairman, British Telecom

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- To families at home, and when divided by distance.
- To politicians and other entertainers.
- To innovators, entrepreneurs and, lest we forget them, to regulators.

This is also a challenge that faces the people in British Telecom, indeed, a challenge that faces our whole industry. I believe that it is a challenge that runs deep into our society; it is a challenge that just about anyone here can play a part in answering.

In British Telecom we often remind ourselves that, if the technology makes something an economic proposition and customers want it then, sooner or later, the regulator will have to allow it. Of course, we do live in a world where competition is regulated, nationally and, nowadays, internationally.

Modern communications have created fast moving markets. They have made global competition a reality; they have made it possible for ideas and information to travel at the speed of light even if the individuals who are communicating with each other are stuck in traffic jams thousands of miles apart.

We may not be able to beam you out of those traffic jams but the latest means of communication, once the stuff of science fiction, are now seen as essential. Portable telephones are everywhere and portable computers and fax machines are coming into wider and wider use.

GROWTH IN COMPETITION

The First Lord Nelson of Stafford saw telecommunications in the United Kingdom being gathered into a nationalised industry based on an economic theory which saw telecommunications as a natural monopoly.

When he died in 1962, the telephone service in the United Kingdom, except in the City of Kingston-upon-Hull, was headed by a Government minister—the Postmaster-General.

In the 75 years of his life, the UK network grew to 5 million exchange lines; in the 18 years since then it has grown again nearly five-fold. Today we have 24 million exchange lines.

But the recent growth in the number of network operators in the United Kingdom has been just as startling, if not more so. Apart from British Telecom, we have a national competitor in Mercury. We have a lot of potential local competitors with cable TV networks, and we have fierce rivalry between the two cellular operators. There is Phonepoint and its rivals with the simpler, cheaper, cordless telephones and three separate personal communications network groups have just been licensed.

Besides all of these players there are the seven specialised satellite services operators, the paging companies, and at least ten foreign telecommunications operators have already established a presence here in the United Kingdom.

This makes Britain the most open telecommunications market in the world. Have no doubt about it, at British Telecom we believe that competition leads to progress and better service for all customers.

Take the ordinary telephone instrument in the old days; you could rent any colour the GPO chose to provide—provided you did not mind black. Today you have a choice of over 250 different instruments, and most people choose to buy at least one telephone from a competitor of British Telecom. Today you can choose from modern 'antique' telephones right through to those in the image of Mickey Mouse and Garfield.

If an office manager in Lord Nelson's time wanted a medium-sized private exchange, he had only one choice of supplier—the GPO. His modern counterpart has 11 suppliers representing five different nationalities and these suppliers offer 26 separate products.

COMPETITIVE ENVIRONMENT

I mention different nationalities because, in recent years, in the supply industry, there has been a series of changes that have continued to modify the shape of the industry in which Lord Nelson and his son both served. The involvement of Siemens in the restructuring of the UK's telecommunications industry is just one example of a new wave of international realignment.

Take technology for instance. When it is not competition between fundamentally different technical solutions, such as satellites and cables, then it is competition between designs, between standards, or between different suppliers. There is competition between different software for coding, multiplexing, and compression; competition between inventions, opinions and even, dare I say it, between prejudices.

If it is not competition between technologies then there is competition between different owners or different approaches to ownership. We have seen it in space with PANAMSAT and ORION as competitors to INTELSAT; in Europe with ASTRA, as a private competitor to EUTELSAT.

There is competition under the Atlantic as well with two optical-fibre systems (TAT8 and PTAT) providing customers with the opportunity to go for diverse routing.

We are likely to see that same competition under the broader waters of the Pacific with TPC 4 and NPC. It may even come across the Soviet Union with rival cables facing the challenge of the weather of Siberia. In the ether, there is competition for the electromagnetic spectrum.

Conventional fixed point-to-point links, satellites and broadcasting now find competing demands on spectrum intensifying as mobile communications expand, on land, at sea and in the air. Indeed, many people want terminals that can be used conveniently anywhere at any time.

Mutual force reductions amongst the military super-powers may arrive just on cue and could

well free some bandwidth. The handling of the radio spectrum is an important challenge to the nations of the world. Take Skyphone, BT's payphone service in the air. We have had to go to the American Federal Communications Commission (the FCC) to get some sense into the regulatory aspect. Think of British Airways flight BA 217; it flies Heathrow to Washington DC to Pittsburg. With Skyphone fitted, all the way across the Atlantic, it would be quite legitimate to make calls via an INMARSAT satellite.

But the problem is whether one can legitimately use the international INMARSAT system—the one used from Heathrow to Washington—for the domestic leg from Washington to Pittsburg, rather than switching to a domestic US monopoly.

That kind of monopoly can only exist through regulation. The satellites themselves are no respecters of national boundaries. In this case, it is the FCC that makes the rules. The monopoly means extra equipment and extra hassle for our customers.

These are not matters uniquely for the FCC. They are not simply domestic matters for the United States. They are not even just squabbles between the United States and the United Kingdom. What we are talking about are matters of regulatory competition: matters in which other United States agencies—the State Department, the Department of Commerce, and the Federal Aviation Agency—take a lively interest. They are matters which the politicians on the Ways and Means Committee on Capitol Hill are beginning to watch carefully; matters which concern our politicians here in London too and the European Commission in Brussels.

We find ourselves in a technical competition that does not take place on a level playing field marked with clear straight lines. Our game takes place amidst a veritable maze of regulatory hedges and ditches with different rules for different players. And the mazes are different country to country, and region to region.

Here is a challenge for the imaginative engineer, and a fascinating set of questions for strategic thinkers.

But regulatory competition is often a different name for national competition, competition between governments, between political ideologies and between peoples, for wealth and for power.

We are talking about competition between States, including their telecommunications players, for national advantage.

The desire for national advantage still cuts across the increasing globalisation of markets. It is true in the globalisation of financial markets, telecommunications, and in trade for other goods and services. It even applies to sport and entertainment.

We have major multinationals and transnationals bridging boundaries and making a nonsense of many aspects of nationalism.

But one cannot avoid noticing that the home countries of some of the biggest multinationals (the United States and Japan) are tussling hard even when Japanese companies are trying their best to follow the example of IBM in becoming good local citizens.

Telecommunications is still high on the political agenda in every major country, and with good reason. Strategic advantage based on military power may be waning, but strategic advantage based on good intelligence (commercial and cultural as well as political) has never been more vital.

MEETING CUSTOMER REQUIREMENTS

While some people are prepared to sacrifice economic well-being for their own brand of nationalism, a lot of people realise that to meet the wants of their fellow citizens, in an interdependent world, they have got to meet the requirements of consumers, both at home and abroad. Both nationalism and globalisation seem to be intensifying at the same time.

Commercial competition is just as significant as technical competition, regulatory competition and national competition. Today, customers for telecommunications services are all trying to gain competitive advantage over each other.

In telecommunications, we remember that man called Strowger who invented the automatic telephone exchange, controlled by a customer dialling in a series of numbers. Strowger was an undertaker in Kansas City. He was losing business to his competitor because his competitor's wife was the telephone switchboard operator. She was diverting calls from the bereaved straight to her husband.

Let us focus on Strowger's competitor. Was he not a classic example of the imaginative entrepreneur? Did he not gain competitive advantage through introducing intelligence in the network? And what about Strowger's answer? Was it not to devise an open network? Strowger developed a network that allowed prospective customers to bypass the 'value added' service provided by the network operator. At last, customers could get the circuit they wanted without delay, without the cost of a human operator and, at least in theory, without the risk of diversion.

Coming closer to home, Reuters must be one of today's largest users of international telecommunications. They want to be sure that nobody gets news of financial deals signalled to any city in the world faster than they do. Reuters want that information delivered within a second.

Satellite broadcasting gives convenience and flexibility. But the signal has to travel all the way out to the satellite and back. That takes time even at the speed of light, but submarine optical-fibre cables also offer communication at the speed of light and give the receiver of the data a head start.

Timely, accurate, focussed information is an essential weapon in the competitive armoury.

Another example of this is calling-party identification, where the number of someone telephoning you is shown on the display on your telephone instrument. In Britain, we have tended to think about this in terms of catching people who make nuisance calls. In America, calling-party identification allows well-equipped companies to respond immediately with their customers' name and details to give a personalised service.

That personal touch, the personal approach to customers, makes a real difference. We have to ask who? as well as what? when we ask about our customers' requirements in the 1990s and beyond.

THE COMPETITIVE CHALLENGE

It is maybe in this area that we find the reason why we are celebrating Lord Nelson of Stafford: his leadership went far beyond engineering. Lord Nelson demonstrated the importance of paying attention to people and to the contractual arrangements that governed supply.

This is what quality management is all about. 'Quality' means meeting individual customer's agreed requirements at the lowest cost, first time every time. That is the contract. For me, delivering that quality service represents the greatest competitive challenge British Telecom faces in the world communications markets.

It is not going to be enough to have imaginative engineers, working in isolation. We must have teams who are capable of listening, capable of responding to needs of other people, capable of collaborating across the barriers that too often divide.

British Telecom is undertaking a fundamental restructuring in its management. We are building an organisation of teams, structured to meet individual customer requirements in place of one based on product and geography. We are bringing the engineering and management of our networks, whether in the UK or abroad, into a single, coherent, whole.

I have an ambition for British Telecom. Our ambition is to be the most successful telecommunications company world-wide. Not necessarily the biggest, but to be the telecommunications company everyone of you would acknowledge to be pre-eminent, to be the one you turn to naturally if you have need of a world-wide network.

This is our challenge in the world communications markets of the 1990s. To meet it we shall have to break down barriers: the barriers established by national politicians, protecting their domestic markets; the barriers maintained by domestic regulators, protecting their national politician; the traditional barriers between the engineering profession and the accountants and marketeers—the barriers between the technological concept and the practical delivery of value for money in the market-place.

Above all we need to breakdown the barriers between ourselves and our customers. To meet our ambition, the challenge is to understand the communications' needs of those customers better than they do themselves and to provide them with quality solutions, tailored (or should I say engineered?) to meet those needs.

This is a far cry from what Lord Nelson of Stafford in his day would have recognised as the challenge facing the GPO. But a challenge in which I feel sure he would have revelled, as, indeed, we are revelling ourselves.

Biography

Iain Vallance was appointed Chairman of British Telecom in 1987. He joined the Post Office's North West Region in 1966. Two years later he moved to the Post Office Headquarters in London. He has held a variety of posts in British Telecom, initially in finace and procurement, and has been on the Board since its inception in 1981. At the outset, he was responsible for organisation and business systems, and subsequently became Deputy Managing Director Inland Division. He was made Managing Director Local Communications Services Division in 1983, and in October 1985 he was appointed to the new post of Chief of Operations. He became Chief Executive in October 1986 and Chairman on 1 October 1987. He is a member of the CBI President's Committee, the President's Committee and Advisory Council of Business in the Community, and the Governing Body of the London Business School. He is Chairman of the Foundation for Education Business Partnerships. He was educated at the Edinburgh Academy, Dulwich College, Glasgow Academy and Brasenose College, Oxford. He graduated with a B.A. in English from Oxford in 1965 and an M.Sc. from the London Business School in 1972. He was elected a Fellow of the London Business School in 1989.

Voice Processing Systems in British Telecom

R. E. WALTERS†, and K. R. ROSE*

This article provides an overview of voice processing systems with particular reference to their existing and potential uses within British Telecom. It addresses voice messaging, interactive voice response and audiotex and is based upon a survey conducted within the business in 1989.

INTRODUCTION

Late in 1989, an extensive survey of the existing and forecast use of voice processing systems within British Telecom was conducted by two members of the Communications Systems Division (CSD). The survey was initiated by the cross-divisional speech processing group of CSD which included representatives from all the relevant parts of British Telecom. The group existed to improve awareness of speech or voice processing activities across the business, to stimulate and track standards in this area and to identify and initiate developments of common interest. This article provides an introduction to voice processing systems, the technology used within them, the market for them, the standards situation and of course BT's current and future use of these systems. In doing so it provides a useful, but by no means in depth, overview of the voice processing world and its relevance to BT based on the survey's findings.

Voice processing has many definitions; during the work of the cross-divisional group and the survey, the definition adopted was that generally recognised in the United States, where the market for this technology is already mature. Figure 1 provides that definition and is based essentially on a systems viewpoint of voice processing.

As can be seen, the three main areas covered are voice messaging, interactive voice response and audiotex. These are explored in more detail in the next Section; it is sufficient to say here that, though the underlying technology of all three areas is similar, the market, and hence the marketing approach to each, is very different.

As will be seen, BT is already active in the voice processing area across the whole piece. Voice processing is an integral part of offering complete telecommunications solutions and is rapidly becoming more so.

† Customer Systems, British Telecom Communications Systems Division

* Speech and Language Processing Division, British Telecom Research and Technology

VOICE PROCESSING SYSTEMS

As mentioned in the introduction, voice processing can be divided into three broad functional application areas (see Figure 1). In this section, the functional operation of these application areas is described.

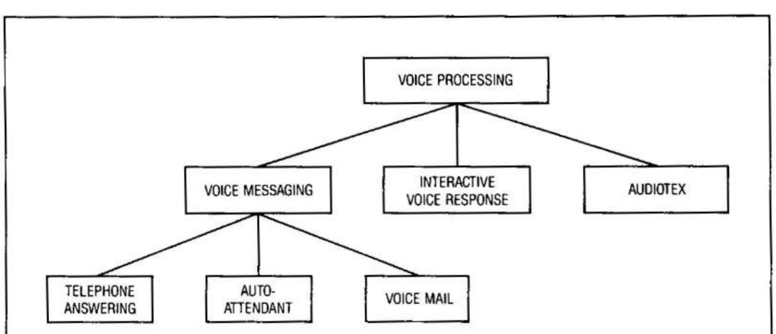
Voice Messaging

There are many products currently on the market that address each of the three areas under voice messaging; that is, telephone answering, automated attendant and voice mail. However, some confusion exists as to where the dividing line between these areas lies. For example, voice messaging and voice mail are terms often used interchangeably, whereas, in practice, voice messaging systems invariably offer all three of the areas mentioned. Therefore, it is worth exploring each application to appreciate these differences and assist in understanding the operation of a voice messaging system.

Telephone Answering

Telephone answering is by far the most simple and best known function carried out by a voice messaging system. As the name implies, a telephone answering system simply answers telephone calls and invites callers to leave messages. Single-line telephone answering and recording machines (TARMs) have been in use for a number of years and there are many on the market. Telephone answering on a voice messaging system is effectively a multi-user TARM.

Figure 1
Definition of voice processing



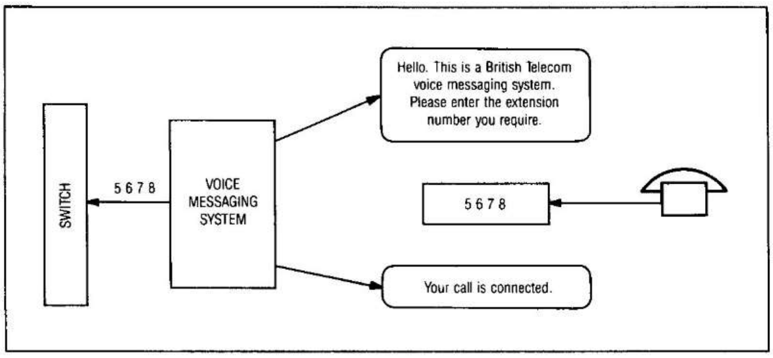
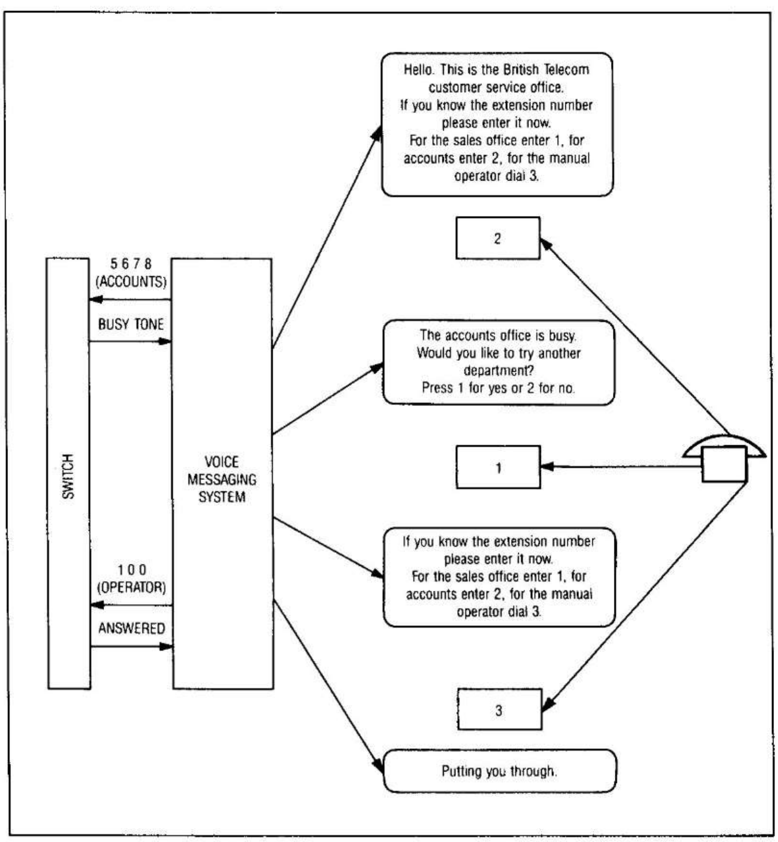


Figure 2
Simple automated attendant operation

Owners of single-line TARMs use a personalised greeting inviting callers to leave a message. However, if a voice messaging system is connected as a centralised service on a PABX, as is often the case, there will be many different calls diverted to it. If the original destination of a diverted call is sent to the voice messaging system from the PABX, then it is possible to offer callers a personalised greeting from the owner of the answering mailbox to which they have been diverted. If the voice messaging system is not informed of the original destination, then callers receive a general greeting and all messages are stored in a public answering mailbox or the caller is again asked to identify the person required.

It should be apparent that informing the voice messaging system of the original destination of a diverted call substantially improves performance from both a caller and owner perspective. If this information is available to the voice messaging system it is said to be *integrated* with the switch. The example given is of the most simple form of integration. The integration can become more sophisticated if additional call status information is sent to the voice messaging system. For example, a different greeting can be given depending on the reason for the call being diverted.

Figure 3
Sophisticated automated attendant operation



Automated Attendant

The role of an automated attendant is to imitate the operation of a human switchboard operator. It would have been logical therefore to call this service *automated operator*; however, like a number of similar services, it was imported from the USA where operators are known as *attendants*.

An automated attendant system is usually placed on the primary answering point of a telephone switch. For example, in a PABX this could be on one of the operator circuits or a designated extension. The automated attendant answers incoming calls and gives an announcement to the caller inviting him/her to enter the extension number of the person he/she wishes to speak to. The caller then transmits the appropriate digits using the telephone keypad (loop disconnect or MF4 signalling) or speaks the digits (which are detected by a speech recognition system). The automated attendant then connects the caller and drops out of the call. This simple interaction is shown in Figure 2.

The operation described above is very simple and falls some way short of emulating a human operator and would be of little practical use. For example, it assumes that the extension given by the caller actually exists!

In practice, automated attendants are much more sophisticated and will endeavour to ensure that all calls are successfully terminated. Most implementations will offer directory searching and simple departmental searches (for example, Key 1 for sales, 2 for accounts, and so on). The dialogue shown in Figure 3 is a more realistic example of an automated attendant which is integrated with a switch.

The potential of being able to replace a human with a machine makes an automated attendant very attractive. Apart from the obvious cost savings, the automated attendant works 24 hours per day and results in more calls being answered. As it enables callers to get direct access to extensions, it is often seen as replacement for the rather expensive service of direct dial in (DDI) that is implemented on modern switches. For this reason, automated attendant is often crudely referred to as *poor man's DDI.*

Voice Mail

As the name suggests, voice mail is the speech equivalent of electronic mail. In fact, the facilities offered by both voice mail and, for example, Telecom Gold are almost identical.

A voice mail system is a collection of individual mailboxes, each capable of holding a

number of voice messages. From a caller's point of view, voice mail is similar to a telephone answering and recording machine. However, the owner of a mailbox has a range of sophisticated facilities to service the voice messages in the mailbox and these are clearly differentiated from answering machine functions.

Most systems offer a large number of facilities. Apart from the basic operations of listening to, saving and deleting messages, the facilities available are similar to those that can be performed when servicing paper mail. However, to highlight the operation of these systems some of the more powerful facilities are described below.

Reply If the sender of a voice message also has a voice mailbox, the recipient is able to reply without having to know the sender's mailbox number.

Forward A further voice message can be appended to a voice message and forwarded to a third mailbox owner. The recipient of this message is then informed of the source of both messages.

Broadcast Perhaps one of the most powerful facilities. A voice message can be simultaneously sent to several predetermined mailboxes.

Interactive Voice Response (IVR)

An interactive voice response system (IVR) is the voice world's solution to remote computer access via the telephone network. The IVR is connected between the telephone network and a computer and effectively emulates the function of a terminal. See Figure 4. Where a keyboard is used to enter commands on a terminal, either a telephone keypad or spoken word is used to enter commands into an IVR system.

The IVR enters into a dialogue with callers and extracts the appropriate response to form a computer command. When the command is complete, the IVR accesses the computer, converts the retrieved data to speech and sends it to the caller.

Audiotex

Audiotex systems are the juke-box of the voice world. They are sometimes referred to as *voice publishing systems*. These systems store a range of voice announcements, such as weather forecasts for different regions, and are accessed via the telephone network. In the simplest case, the information required is given a specific telephone number; in more sophisticated applications, callers to the system are asked to select the service they require by using either the telephone keypad or spoken words. An example of such a dialogue is shown in Figure 5.

In some applications, callers are invited to leave voice messages at certain points in the dialogue and so audiotex machines gradually begin to merge with voice messaging and response information.

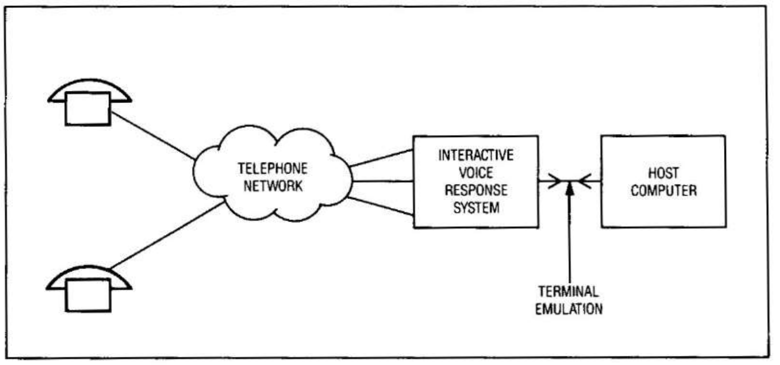


Figure 4
Interactive voice response system

VOICE PROCESSING SYSTEMS PLATFORMS

At this point, it should be fairly apparent that the systems so far mentioned are either announcement machines, recording machines or a conglomeration of both. It should therefore come as no surprise to know that systems can be purchased that perform one or more of these functions. Not only does this mean a saving in both cost and space, it also offers operational advantages. For example, a voice messaging system attached to a PABX would greet callers with an automated attendant service. If it cannot complete the connection, it invites the caller to leave a message. The message can be left in the mailbox of the person called. If they do not have a mailbox, the system enters telephone answering mode and invites the caller to leave a message in a public mailbox ensuring that the name of the recipient is included in the message. The public mailbox is then reviewed by an operator at a later time and the stored message can be passed to the intended recipient when

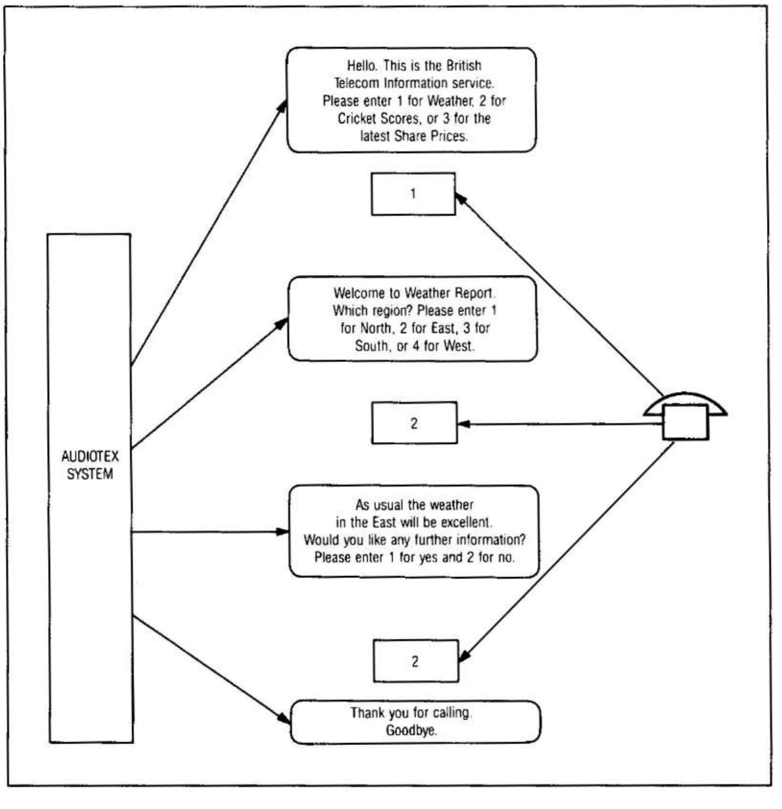


Figure 5
Audiotex operation

available. In this example, the voice messaging system keeps total control of the call and ensures that all calls are answered. Every call is either successfully connected or the caller is invited to leave a message. This operation is commonly known as *call completion*.

The success of voice processing systems is linked to two aspects: the degree of integration with the host switch (as discussed in the section on telephone answering) and message waiting notification (that is, when the system notifies a person that it has a new voice message for him/her). Without message waiting indication, a mailbox owner either forgets to review messages in the mailbox or gets totally disillusioned with the system when continually informed that there are no new messages after having accessed it. Message waiting indication can take a number of forms, from simply ringing the telephone of the message recipient at periodic intervals, to lighting a dedicated lamp attached to the recipient's telephone.

Voice processing systems are available to-day that offer both voice messaging and audiotex. Interactive voice response systems tend to be more application dependent and are therefore often bespoke.

Since the basic operations of the applications described so far are the same, it is logical to assume that they are architecturally similar and use the same technology.

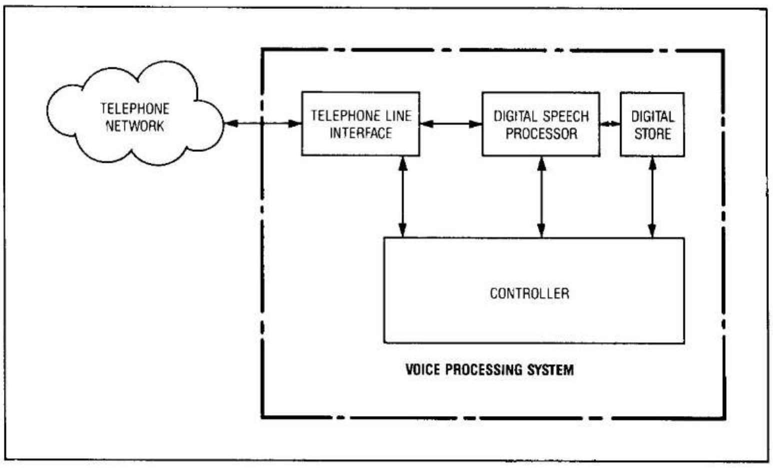
VOICE SYSTEM PROCESSING TECHNOLOGY

Fundamentally, voice processing systems consist of a few basic elements: a method of interfacing to the telephone line or lines; a means of processing the voice information from this interface for storage or retrieval; a suitable storage device; and a controller for all the elements. Figure 6 presents the interconnection of these elements.

Naturally the degree of sophistication of the elements themselves and their interconnection varies enormously—dependent on application—but the basic architecture is still discernible.

The function of the telephone line interface is to terminate the telephone line. Control information is sent to the controller and can be both commands from the caller (transmitted using the keypad or speech) and call status information, such as diversion type. Speech, which normally exists in analogue bi-directional form on the telephone line, is digitised and presented to the digital processing element as separate transmit and receive channels. Digital processing of the speech signal can provide a number of functions, the most common is that of compression and its reverse in order to reduce the bit rate of the digitised speech and therefore economise on the amount of storage required.

Figure 6
Voice processing system



The digital store can take many forms; its function is to provide storage for speech files, which may be voice messages, pre-recorded announcements, segments of speech for concatenation into speech responses and so on. Additionally, the digital store holds control programs and data relevant to the voice processing system functionality.

The controller is a data processing element which controls the movement of stored information in response to commands from the telephone line. The controller may be connected to a computer terminal, an entirely separate computing system or to a PABX. It can also interpret call-related information from the telephone line interface and control the set up, clear down and progress of calls. In certain applications, it communicates with the digital speech processor to manipulate its functionality or to receive information.

One simple view of the general architecture of voice processing is to regard them as computers which possess a rather unusual input/output port—the telephone line.

An example of a telephone line interface and digital speech processor is the BT speech card developed in the Speech and Language Processing Division at the British Telecom Research Laboratories (BTRL). It is designed to plug into an IBM-compatible Personal Computer (PC). The PC performs the function of digital store, storing the speech on its hard disc, and controller.

FUTURE TECHNOLOGY

To consider the future technological developments in voice systems, it is worth considering their current limitations and how they can be overcome.

By far and away the most serious present limitation for voice processing systems is the inability for a caller and user, for example, the owner of a mailbox, to remotely control the system. The vast majority of systems use DTMF signalling for this purpose; however, the penetration of this type of signalling in the network, especially in the residential area, is relatively small. Very few systems use loop-disconnect signalling which leaves speech recognition as the only realistic alternative. Isolated word speech recognition systems are sufficiently accurate to offer callers remote control of systems. Recog-

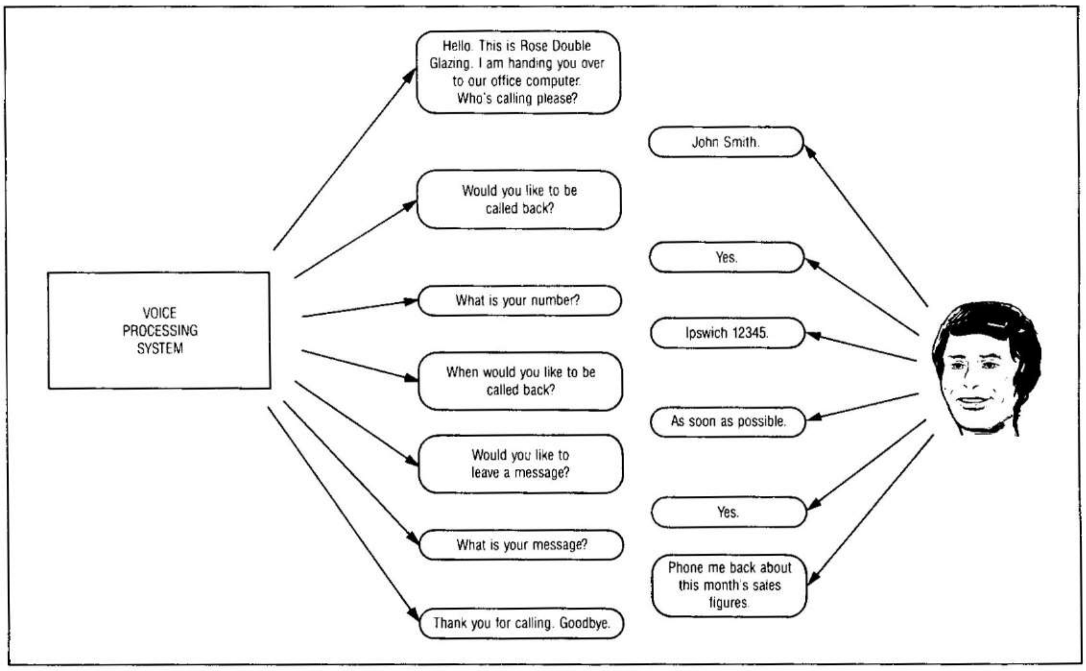


Figure 7
Interactive dialogue

nisers are now becoming available which are able to detect connected words which will further improve user control.

A second area in which all current systems are limited is the human interface, from both a caller and user perspective. Any owner of a TARM, especially in the UK, would highlight the fact that the number of messages left on his/her machine is much lower than the actual calls received; that is, callers have not left a message when they have encountered the answering machine, a phenomenon often referred to as *slamdown*. There appears to be a pathological fear among British people to talk to machines and while this continues the take up of voice processing systems will be slow. One method that promises a long-term solution to this problem is for the machine to enter into a dialogue with the user instead of giving a single greeting and inviting a single response. An example dialogue is shown in Figure 7. The objective is to ask a simple question that the caller can answer without much thought. If after the caller's name, telephone number etc. have been extracted the caller does not leave a message, the owner of the machine can still return the call.

From a user perspective, two limitations of voice processing systems are: having to remember codes for invoking facilities; and message waiting indication. Once again, speech recognition can be used to avoid using facility codes; it is a lot easier to remember the name of the required facility than its associated code.

More sophisticated integration can also improve the interface. For example, more powerful facilities and easier invocation can be achieved by using the D-channel of an integrated services digital network (ISDN) connection.

Other drivers in speech technology include:

(*a*) the progressive integration of voice processing systems with data processing and telecommunications systems;

(*b*) the provision of friendly, fast voice application generation tools; and

(*c*) the combination of advanced voice recognition solutions with artificial intelligence to provide context driven recognisers able to respond to natural language.

Finally, existing speech synthesisers which convert text into voice are difficult to understand which has meant their current application in interactive voice response systems has been limited. As the speech quality produced by synthesisers improves, interactive response systems will be used in increasing numbers as more and more applications are tackled.

VOICE PROCESSING STANDARDS

With the possible exception of speech encoding there is a dearth of standardisation in the voice processing area. Even in the speech encoding area, standardisation has lagged the market considerably and there are already a vast number of systems utilising non-standard compression techniques. The areas where standardisation is required are as follows:

(*a*) interchange of voice messages between systems,

(*b*) speech encoding algorithms,

(*c*) integration of voice messaging systems with text messaging systems,

(*d*) integration of voice processing systems with switches,

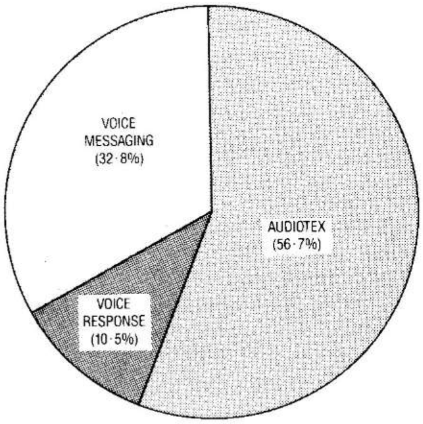
(*e*) common user interfaces particularly from the telephone keypad,

(*f*) common numbering and addressing, and

(*g*) common responses and dialogues.

The major activity so far has concentrated on the passing of voice messages between voice mail systems supplied by different vendors. A sub-group of the Information Industries Associ-

Figure 8
Market breakdown in US

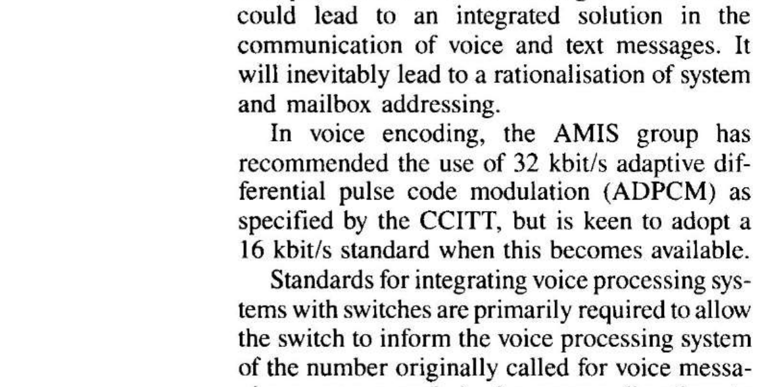


ation in the United States has formulated a protocol for this purpose called the *Audio Message Interchange Standard* (AMIS). Although this has no international standing, it is based on X.400 and has the backing of a number of significant vendors and users in the United States. BT has some involvement with this activity through its minority shareholding in VoiceCom which participates as a user representative. A similar group has now begun work on standardising the user interface. The CCITT itself has now begun work on providing extensions to the X.400 definition to encompass the transportation of voice messages. This work could lead to an integrated solution in the communication of voice and text messages. It will inevitably lead to a rationalisation of system and mailbox addressing.

In voice encoding, the AMIS group has recommended the use of 32 kbit/s adaptive differential pulse code modulation (ADPCM) as specified by the CCITT, but is keen to adopt a 16 kbit/s standard when this becomes available.

Standards for integrating voice processing systems with switches are primarily required to allow the switch to inform the voice processing system of the number originally called for voice messaging purposes and, in the reverse direction, to provide a message waiting indication. This is a notoriously proprietary area with each switch having a specialised interface. The computer-supported telephony standard currently being evolved by the European Computer Manufacturers Association (ECMA) is likely to provide a basis for a common interface to PABXs. In public switching, the various activities in standardising the intelligent network are likely to provide a similar solution, though in the longer term.

Figure 9
Voice processing applications within BT



The area of common responses and dialogues has not yet been addressed by standards groups; however, it has been suggested that this, and a suitable user interface definition, could be included in a future issue of the BT Product Style Guide.

VOICE PROCESSING MARKET

At the end of the last decade, there were said to be over 50 000 voice processing systems in operation in the United States but in Europe less than 500. Analysts give various reasons for this very marked difference, but generally agree that the European market lags that in the US by approximately 4 years.

Major growth in voice processing is therefore anticipated, though for many reasons it is predicted that the European market will never be quite as large as that in the United States. The first voice mail system was installed in the US in 1980, but the major growth has occurred in the last 4 or 5 years.

Revenue split in the US for the three major sectors identified previously is presented in Figure 8.

In the UK, the audiotex sector is undoubtedly much more dominant. Call revenue from CallStream has grown from £2M to £130M per annum in the past 4 years and is expected to double over the next 5 years.

The voice messaging market in the UK is, in comparison, significantly smaller at present. It is dominated by bureau-supplied voice mail services (such as BT's Voicebank), but the number of privately owned systems and networks is growing. There are currently less than 100 000 public mailbox users though this is predicted to quadruple in the next 4 years. Currently, there are less than 200 private voice mail systems in the UK, but once again this is expected to rise significantly to 1400 by the year 1994.

The market for interactive voice response systems is certainly growing at present. Many of the major banks, building societies etc. are currently installing systems.

Overall the picture is a rosy one though it should be noted that many of the predictions for growth in this industry in the 1980s have not been realised outside of the United States. The general picture of the market is of increased competition from suppliers, particularly US-based vendors moving into Europe, and a slow but growing awareness of the benefits of voice processing systems by customers.

BT's CURRENT USES OF VOICE PROCESSING SYSTEMS

BT has a large number of voice processing systems in operation, the number and the overall coverage in terms of varieties of applications is growing annually. Figure 9 provides a useful

mechanism for viewing the use of this technology both within the company itself and as a component within customer sales.

This review is conducted from the standpoint of gaining an appreciation of where and how voice processing technology is used across the whole of BT. The diagram is therefore arranged around the nature of the applications within the business rather than the technology itself. Of course the same technology will, in many cases, be capable of providing most of the applications which are discussed though choice of equipment will be strongly influenced by the specific requirements of that application; for example, its traffic capacity.

The major split shown in the diagram is between cost reduction and revenue attraction. These are considered separately.

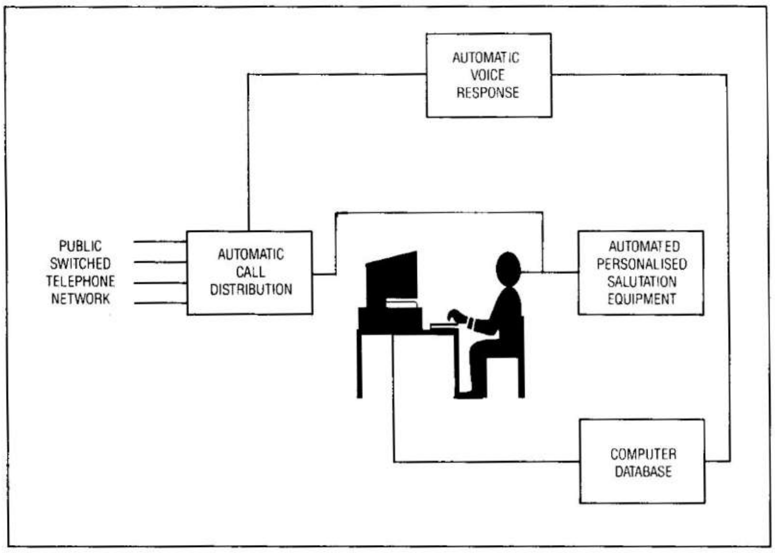


Figure 10
Directory enquiries system

Cost Reducing Applications

These are the applications of voice processing which do not result in the sales of services or equipment but can significantly reduce operating costs. The obvious target for using voice processing in this way is for automating functions which are currently carried out manually. This is, of course, an on-going trend which began the day that Almond B. Strowger conceived and implemented the first automatic telephone exchange. Though Strowger's innovation eliminated the need for vast numbers of telephone operators, there were still many functions which required, and still require, the presence of a human operator. The most notable of these is the directory enquiry service, but there are a number of others ranging from advice of call duration to the old-style FreeFone service.

The directory enquiries application both within the UK, and to a lesser extent internationally, is of most interest. It has been estimated that savings of around £3M per annum can be made for every second that is shaved from the existing operating time. By using an automatic voice response system, that is, the number is spoken to the enquirer automatically, and by the use of automated personalised salutation (whereby the operator is relieved of the burden of greeting the incoming caller), a total of around 8 seconds can be saved on each call. Figure 10 gives an overview of what is being implemented as a trial within BT at the moment.

As can be seen, the implementation requires an automatic voice response system to interpret the number delivered by the directory database equipment. This assumes connection to the front-end automatic call distribution switch so that, when the operator and incoming caller have decided upon the correct entry, the call can be diverted to the automatic voice response equipment and the number delivered. The automated personalised salutation equipment does not require any integration into the system; it effectively appears in the operator's microphone lead.

Such systems have been in use in the United States for nearly a decade; therefore, after the current trial, early implementations of this voice processing application can be expected within BT.

One other example of the automation of manual services is the use of a voice system to allow cashless calling from payphones. Obviously, this can be done through a human operator but, since all public payphones are capable of sending DTMF digits, an interactive voice system can perform this function very effectively. In this case, the voice system asks questions and the caller provides identity codes etc. via use of the DTMF keypad.

Note that one of the interesting things about the introduction of voice systems into the network is that installation can be gradual and need never be total. A human operator backup can always be maintained and most voice systems do give 'breakout' options to allow a human operator to take over the call.

There are a number of related operator services that will benefit from these simple voice systems, but these will not be discussed here.

One other application area in which BT's operating cost can be reduced is in the use of voice systems for interrogating central databases from a telephone. The Caeser equipment currently in use in the field is a very good example of such an application. Here, voice equipment enables an engineer using very simple instructions to access a database to, for example, find a free cable pair. The same equipment, with minor reconfiguration, can be used to access almost any database. Such systems can be used quite effectively for scheduling work, thus obviating the need for a central dispatch desk.

Added to all this there is of course the possibility of using BT's own services to reduce costs. Many salesmen etc. utilise the Voicebank system for call answering especially when the service is coupled to a pager for call waiting indication. Currently, Voicebank is the internally recommended voice messaging solution within

BT. Similarly, BT utilises modern recorded information techniques to store and distribute the many announcements required to satisfactorily operate a telephone network.

Revenue Attracting Applications

As indicated in Figure 9, there are quite a few possibilities here, and these are examined separately.

Network Stimulation

One of the key points to realise about voice processing systems is that almost all of them do stimulate network utilisation either by completing a call where it would otherwise have been uncharged or by creating the need for a call which otherwise would not have been made. The initial justification for developing the interactive banking system now being supplied to the Royal Bank of Scotland was simply call stimulation. The equipment is a voice response system which can be driven by DTMF tones or directly by voice. The system runs on IBM-compatible PCs and is based upon the BT speech card. The sort of transactions that can be carried out vary between requesting a new cheque book to actually transferring funds from your own account to another. Obviously, performing these transactions via the telephone network increases call revenue; it also benefits the bank because tellers and clerks are not required for these tasks.

There are many other applications of speech processing systems which will stimulate call revenue. In fact, some predictions indicate that this will be the major revenue generator of all the applications of voice processing. Future applications will be explored later.

The discussion so far has concentrated on the conventional telephone network. It is worth noting that the income from the derived services network is very significant and growing rapidly. CallStream income is primarily derived from audiotex equipment providing voice information at premium rates. The audiotex equipment may be managed by BT, by an agency or supplied directly by the service provider.

Similarly, Linkline Messagelink provides voice messaging capabilities for 0800 callers thus completing calls which would otherwise fail.

Value Added Network Services

There are already a number of voice-based bureau services; the most well known application is that of voice messaging. Currently, British Telecom operates two bureau voice messaging services; the oldest is Voicebank which provides customers with an individual mailbox number which can be accessed from any telephone. The major use of the Voicebank service is for telephone answering and an option exists for advising receipt of an incoming message via a pager. The system is also marketed via Cellnet as a diversion point for calls which cannot be connected to the mobile telephone.

The second service in the voice messaging area is provided under the VoiceCom International banner. This provides voice mail functions and requires that the sender and the receiver of voice messages has either a DTMF telephone or DTMF sender. VoiceCom International is based upon VMX voice mail equipment which provides all the functions normally associated with a private voice mail system, from simple message receipt to broadcast lists and forwarding. It also has the ability to transfer messages between nodes; therefore, a message recorded on the London system but destined for a subscriber resident on the San Francisco system will be transferred automatically between the nodes.

In the audiotex market, BT not only supplies the lines via the CallStream service but also offers a range of facilities to service providers. At the most basic, this can consist of loading a service provider's data onto one of BT's voice systems based at a remote announcement centre through to the provision of full editorial and recording services for production of the audio information for a service provider via Supercall.

There are also a number of areas where BT itself provides the entire service. A recent example is the Triple Trivia Pursuit game which is one of the few current services that utilises speech recognition to interact with the caller. The Share Call service provides a more serious example of service provision.

Customer Premise Equipment

During the mid-1980s, BT began to market a private voice messaging system supplied by Ferranti. For many reasons, this was not commercially successful and was withdrawn. Market awareness has increased since then and there are now new moves to market a voice mail system for connection to PABXs though there is not yet a system in the portfolio.

The telephone banking interactive voice response system has already been mentioned. This activity was initiated for call stimulation purposes, but is now regarded as a valid product in its own right. It consists of a rack containing multiple PCs plus the previously mentioned BT speech card together with a means of connection to the bank's central computer system. The PCs are linked by a local area network (LAN) within the rack. The system uses both voice recognition and speaker identification, the latter to improve security of access. Though currently programmed to run the banking application, the equipment could be reprogrammed to meet almost any interactive voice response need.

FUTURE APPLICATIONS

During the survey of voice processing, it was evident that there were far more future applications of voice processing than existing ones. Not

all of these will mature into products or services and so this article describes a sample of applications rather than attempting to record them all. Once again, the breakdown provided in Figure 9 is used as a means of categorising future applications.

Cost Reducing Future Applications

It was mentioned earlier that improvements to directory enquiries was the main means of automating manual functions in this area. There are, however, a number of operator-based services that could also benefit from the use of voice processing. The difficulties of supplying a wide- vocabulary speaker-independent voice recognition solution does limit applications. However, the personalised salutation mentioned earlier can of course be applied to normal operator assistance. Advice of charging and reverse call charging are also candidates for automation. In addition, it is likely that the systems implemented within the national operator service will find application on the international operator front, where the multilingual capabilities of voice systems will come to the fore.

Within BT a large number of PABX switchboard operators are employed, and it is likely that auto-attendants will be able to absorb some of this work load.

Future Revenue Attracting Applications

Once again, as in Figure 9, there are three main areas of interest here.

Network Stimulation

There are a number of voice processing applications within the network itself currently envisaged, some of which relate to the intelligent network. Residential voice messaging is currently being test marketed in the United States with mailboxes being rented fairly cheaply ($5–$10 per month). This provides a useful alternative to an answering machine in that calls can be accepted when the line is busy and there is no need for the purchase and maintenance of another item of equipment for the home. The market for this service could be very large indeed. A similar service is also on trial in the US at present called *message delivery*. Here, when callers receive no answer or engaged tone from the called number, they are offered the opportunity to record a message and the system attempts to deliver it for them. Current trials are based on payphones, but voice message delivery could be applied more generally.

The concept of personal mobility, utilising the fixed rather than the mobile network, is very likely to include voice messaging options especially when this can be sensibly integrated into the network.

The audiotex market is expected to continue to expand and become more sophisticated through the use of interaction. Given the relatively low penetration of DTMF telephones it is likely that this interaction will be via voice recognition therefore requiring new versions of the existing voice processing equipment.

Future Value-Added Network Services

One of the most futuristic applications foreseen here is that of language translation. Prototype systems are already available that can perform this function with some limitations.

The voice messaging bureaus are looking towards expanding their interests into interactive voice response as a bureau function. From a technical point of view, this does make sense since the equipment is capable of connection to remote computers. The automation of talking yellow pages and the integration of our current telemarketing operations with voice systems are both under consideration at present. In the wider scene, the whole area of *voice publication* is likely to become an important prospect for BT.

Future Customer Premise Equipment Opportunities

If the American experience in voice messaging and response is to be repeated in the UK then the opportunities for selling equipment to business customers will be very large. Various possibilities are currently being examined including:

(*a*) PABX-based voice messaging,
(*b*) auto-attendant systems,
(*c*) intelligent telephone answering and recording machines, and
(*d*) voice interactive fax bank machines.

CONCLUSIONS

As stated in the introduction to this article, the whole voice processing scene is of fundamental interest to BT; the survey has reinforced this view and added some quantification to it. It will now be appreciated that the voice processing world has considerable depth and breath. Some argue that it is not an industry in its own right but simply an enabling technology. While there may be some truth in this, the American experience certainly indicates the need for a well-focused market strategy and the base technology itself is certainly an area of significant specialisation.

Fortunately, from the technological viewpoint, BT is well served. BT has an entire division of research and development expertise in this area with a solid reputation both within the business and world-wide. BT has people who are well abreast of, and in some cases in the vanguard of, the latest developments. What we now need to ensure is that the marketing, product management and support functions are sufficiently well resourced to ensure that this expertise can solve customer's problems in voice processing.

It is often said that when all is said and done, a lot more is said than done; while this is so voice processing can but prosper.

ACKNOWLEDGEMENTS

The authors wish to acknowledge the help, information and advice provided by the many departments of BT which were active in the speech processing cross-divisional group and those which contributed to the survey. They also wish to thank Andrew Gosden for his major contribution to the voice processing survey which forms the bedrock of this article.

Biographies

Rob Walters is currently managing the Advanced Development Program within CSD. He joined British Telecom as an apprentice in 1963 and his career has spanned public and private switching and transmission through to product development and marketing in voice processing. He was chairman of the CSD cross-divisional speech processing group.

Kevin Rose joined British Telecom as an apprentice in 1973 and spent a number of years working in the Speech Processing Division at BTRL. He left BTRL to work on new product development in the Switching Systems Unit of CSD, where he was also secretary of the cross-divisional speech processing group. He is now head of the Speech Platform Development Group at BTRL.

Digital Cordless Telecommunications—CT2

R. S. SWAIN†

Over the last ten years, the development of cordless telephones in the UK has followed a logical path, from simple analogue instruments designed for domestic and small business use, through to the world's first digital cordless telecommunications system. This progression has been both commercially and technically driven, based on the principle of market maximisation by the selection of cost-effective specifications and implementations. The adoption of digital cordless telecommunications will expand markets beyond those of conventional cordless telecommunications, by encouraging the development of business cordless systems and the public cordless access services (telepoint) that use a common personal handset.

INTRODUCTION

Second-generation digital cordless telecommunications (CT2) is an example of a measured step in technology terms but evolutionary in applications—an acceptable technical risk is balanced against a core technology that satisfies current requirements but opens up many new service and product opportunities. In the following sections, the background, technology, implementation and application issues of CT2 are reviewed and the present position regarding specification outlined.

CORDLESS TELEPHONES—THE EARLY SCENE

At the end of the 1970s, significant quantities of North American standard cordless telephones were sold in the UK. These instruments were not licensed in the UK for they used frequencies allocated to other services and were deficient in terms of speech performance and dialling security, the latter leading to fraudulent use of customers' lines. In response to this clearly undesirable situation, the British Telecom Research Laboratories (BTRL) and Department of Trade and Industry co-operated to produce a national specification for an analogue cordless telephone (CT1) that was competitive in price with the illegal imports, overcame their deficiencies, but, for price reasons, had to use the same technology.

This approach provided the UK market with a worthwhile alternative product in 1983. Presently such instruments sell for as little as £60, thereby driving the illegal products from the market-place, and it is estimated that 1·5 million have been sold.

The first-generation instrument was intended to satisfy the domestic market and in this, and other respects, it can be considered a major success story that has not been matched elsewhere in Europe. However, as a consequence, its inherent quality has tended to gravitate to a level appropriate to this very price-sensitive discretionary market and the basic analogue technology has virtually extinguished any chance of developing new applications for cordless telecommunications and hence products. Those attempts to introduce CT1 into the business world confirmed a need for in-building mobile communications, but also succeeded in demonstrating its limitations of quality, application potential and limited number of radio-bearer channels.

Only eight radio channels were allocated to the service and, as early as 1981, it was realised that a successful sales campaign for CT1 would create a problem of too many cordless telephones chasing too few channels with consequent high co-channel interference levels and possible over-hearing of telephone calls. Clearly a longer-term solution was needed for introduction towards the end of the 1980s.

CORDLESS TELEPHONES—THE PRESENT SCENE

Before the first CT1 was sold, basic research into its replacement was under way[1]—a second-generation digital instrument. The radio regulatory authorities soon decided that the frequency band would be in the region of 900 MHz and in 1980 in-building radio propagation research was started at BTRL to determine whether useful ranges could be achieved at low radiated powers of the order of 10 mW. This low radiated power level was chosen on the grounds of safety and battery economy. This work gave encouraging results[2], but clearly pointed to the variabilities to be expected from one building to another, or more realistically between, say, metal and plasterboard walls, Figure 1[1].

MARKET

Second-generation digital cordless telecommunications (CT2) was, and is, seen as the

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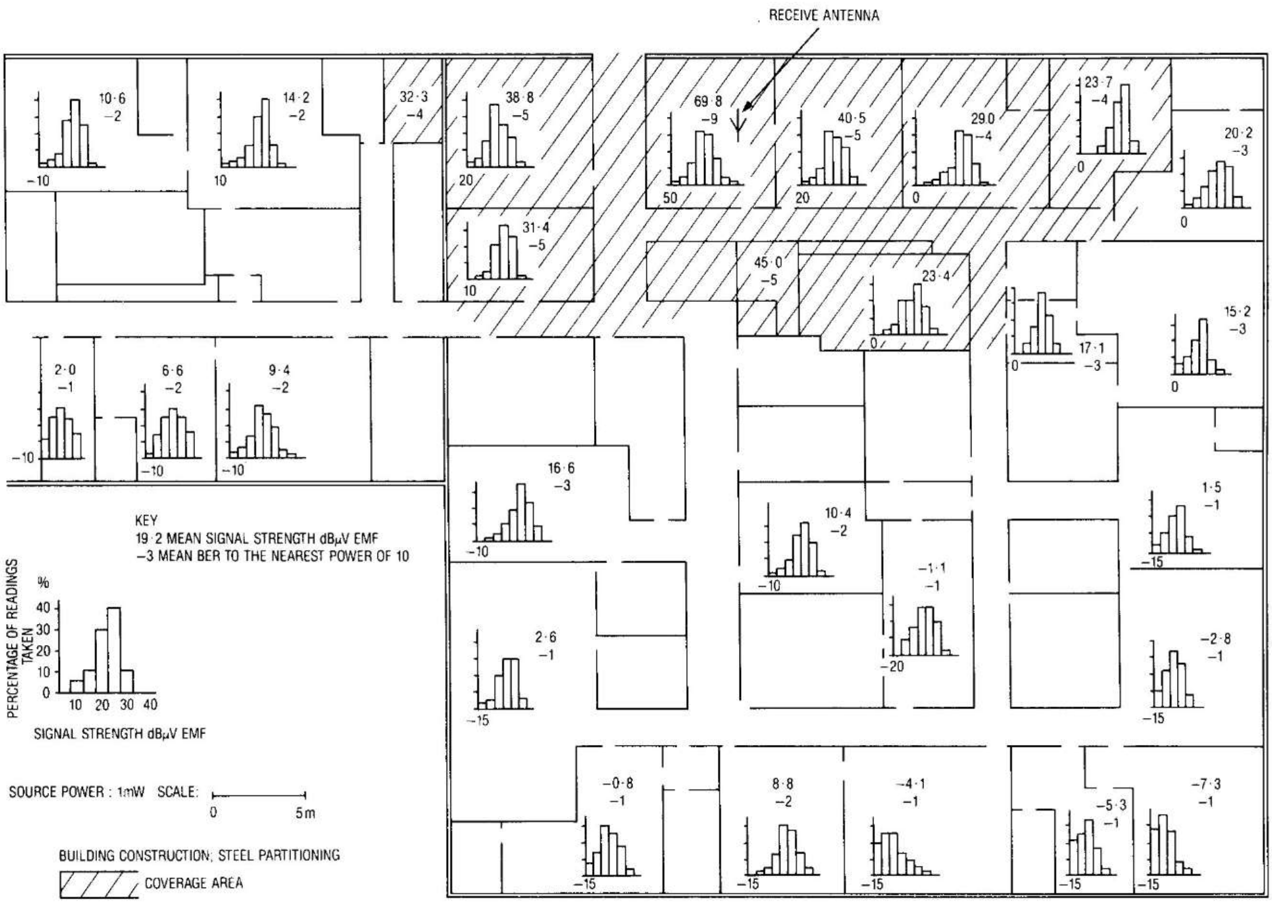

Figure 1
Measured picocell

vehicle to open up the business market to cordless telecommunications, starting with small business applications of the 2 + 4 variety and then expanding upward into multiline multichannel cordless application to PABXs as customer perception of the value of cordless telecommunications grows. Furthermore, as the CT2 unit price falls with time, then the residential market sales expand, and with its increased functionality and robustness to interference will eventually supplement CT1.

Studies of the application of digital cordless communications to the residential and business markets showed average CT densities of the order of 5000 CTs per square kilometre, with a modest 7% cordless penetration of all terminals. But high-rise office blocks with a cordless business communication system have the potential to generate localised densities in excess of 50 000 CTs per square kilometre. It was clear, therefore, that in the absence of large allocations of radio spectrum frequency, reuse techniques were essential to serving the expected demand with a quality and service density akin to that of the wired telephone.

By 1984, studies at BTRL had already shown that the CT2 products about to be developed for residential and business application could equally well be used to provide a public service; the service was known as *telepoint*, Figure 2. The principle of operation was based on a belief that cordless handsets used in the office and home should also be provided with a 'value added' public service from base stations sited in railway stations, airports and shopping centres, etc. Although business and residential use of CT2 offered full incoming and outgoing call facilities, telepoint was limited to outgoing calls only, as each site was connected to the public switched network like a simple telephone or conventional payphone.

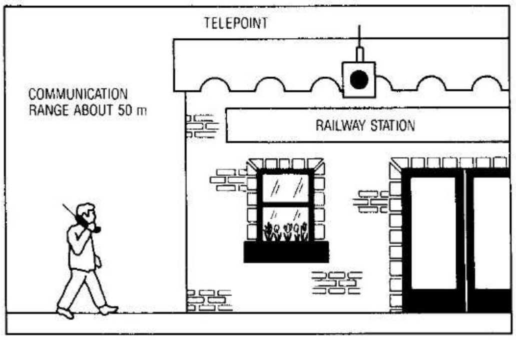

Figure 2—Telepoint

However, the use of a simple network interconnection has kept the cost of telepoint low, and this has allowed the provision of multitudinous access points, sited according to the local traffic demand.

Telepoint, therefore, is not a competitor to wide-area cellular mobile radio, because these services offer incoming call national roaming and automatic call handover while moving between cells. Consequently, telepoint with its cheaper calls is expected to serve the lower echelons of business together with the mass

consumer market. The potential of telepoint was also recognised by the Department of Trade and Industry with the result that, in 1988, four telepoint service providers were licensed to provide service.

The mainstream applications of CT2 are outlined in Figure 3 and from them can be generated a range of sub-applications and products; for example, multiline intercom systems for the office and home, cordless business systems in which the radio control and signalling requirement are integrated with the PBX, a telepoint handset coupled with a wide area message paging receiver to give a quasi incoming-call facility. This last possibility is interesting for it is an early example of converging mobile services and facilities. Of course, being effectively a digital bearer channel, CT2 can have applications connected with data transmission in office complexes, etc.

RADIO ISSUES

The principle of operation of CT2 is shown in Figure 4. Full duplex transmission is provided by transmitting in time-interleaved burst-mode on the same carrier frequency for both directions of transmission. This technique was appropriate for a low-cost instrument since both ends of the radio link are identical. Furthermore, the technique required only a single block of radio spectrum rather than the duplex bands needed for cellular systems. This considerably eased the allocation of radio spectrum as an unused 4 MHz block was available between 864 and 868 MHz. Each cordless channel, therefore, was allocated 100 kHz bandwidth for a two-way speech link. By choosing the CCITT standard 32 kbit/s adaptive differential pulse-coded modulation (ADPCM) speech coding algorithm, it was possible to contain the transmitted symbol rate to 72 kbit/s with some allowance for framing and signalling bits. In practice, the unidirectional B-channel capacity is 32 kbit/s and that for the corresponding D-channel 1 or 2 kbit/s depending on manufacturer choice.

With only 40 channels available to serve the very high traffic demands mentioned earlier, it was clear that the benefits of trunking of the radio channels were needed. Accordingly, the specification of CT2 requires that both handset and base station must be able to operate on any one of the 40 channels. In general, the channel selection algorithm selects that with the lowest interference due to co-channel and/or adjacent-channel interference[3].

Radio Coverage

The combination of radio channel trunking and optimum channel selection enables the 40-channel CT2 system to meet the objective of a mean capacity of 5000 CTs per square kilometre, but more effective frequency reuse techniques must be employed to achieve the ten-fold increase demanded by high-density business systems. Put simply, the problem is to reuse each carrier frequency in the same building, but at the same time making sure that the whole building is covered, otherwise calls will be blocked resulting in a poor performance with respect to conventional wired telecommunications. To achieve the necessary degree of frequency reuse, cordless business systems will have to adopt the cell-coveraged techniques used by the present wide-area cellular mobile systems. The difference, however, is that in multi-storey buildings the cells have three dimensions and critical allowance must be made for radio signal attenuation through floors and walls[4]. In general, for a given building and number of channels, the higher the cordless traffic density (in, say, erlang/km^3), the more base stations required to provide an acceptable channel reuse; that is, coverage from each base station cell gets smaller. Adjusting base station numbers does not of course guarantee coverage into all corners of a building and so positioning of the base stations can be critical. To cope with difficult coverage problems, an alternative solution is to use a radiating coaxial cable rather than an antenna. In this case, the cable radiates for about 6 – 10 m broadside, but it can be bent around corners and corridors, Figure 5.

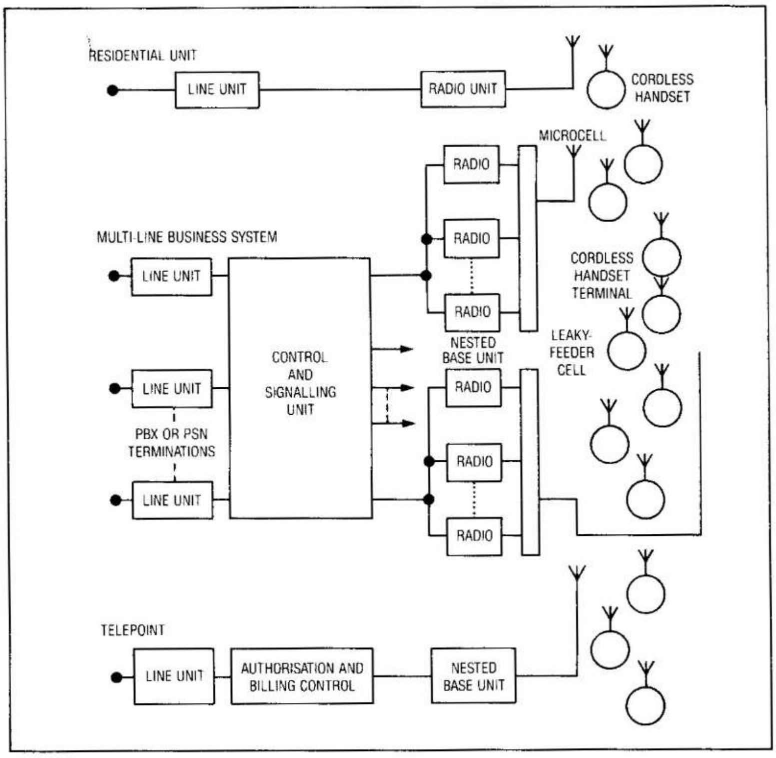

Figure 3
Applications of a cordless telephone system

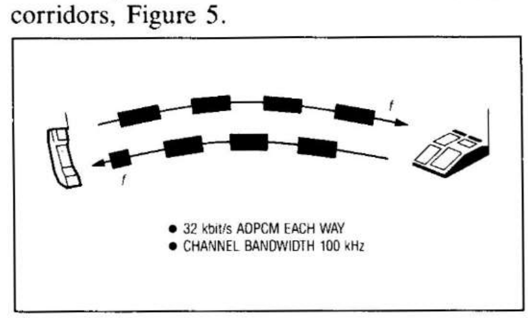

Figure 4
CT2 operation

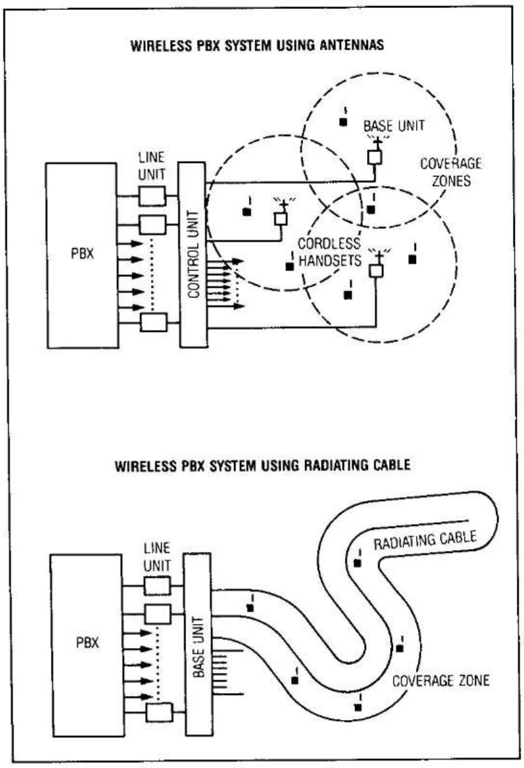

Figure 5
Multiline business systems—coverage options

Implementation

The previous section showed that the CT2 system is well matched to the capacity and service needs of the market and spectrum availability, and yet offers significant opportunities for minimising the inherent cost of the core digital transmission link. For example, the 100 kHz channelling has no need for time-dispersion equalisation, even though there is normally a multi-path radio environment; permits easing of frequency tolerances towards 1 part in 10^5 and allows the use of low-cost radio-frequency filters. There is no necessity for a radio-frequency duplexer, less carrier-frequency sources are required and its moderate processing delay raises no echo control or absolute delay network issues. With these attributes, the radio-frequency circuitry is simplified and the balance of implementation complexity shifts to baseband which is amenable to implementation in low-cost low-power-consumption CMOS and very large scale integration technology; an example is given in Figure 6.

There is also a high-level of core technology common to all applications resulting in high volumes of components and sub-assemblies. This ability to target a large product range using common hardware with inherent cost advantages compatible with the potentially massive residential and business markets is considered to be the key to realising large volume production and, hence, significant economies of scale.

CT2 COMMON AIR INTERFACE

The UK regulatory specifications for CT2 are BS 6833[6] and MPT 1334[7], but these were designed to encourage the maximum degree of innovation and speed in the development of products, and did not force the manufacturers to use a common standard for the parameters of the radio connection between the portable handset and the base unit. Consequently, several different proprietary air interface standards emerged. This was advantageous in the early days of CT2 development, as it allowed individ-

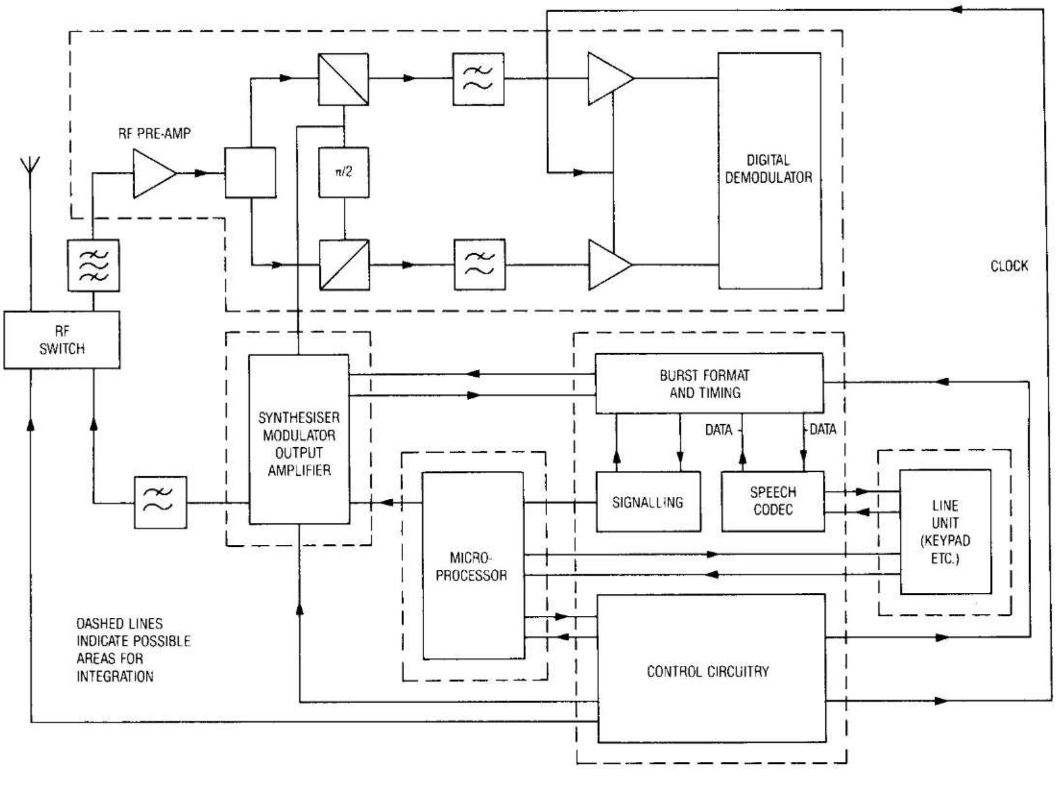

Figure 6
Possible integration of CT2

ual products to be specifically tailored to the needs of their target markets.

The various applications shown in Figure 3, however, clearly point to a need for all handsets, irrespective of manufacture, to be built to a common standard. Thus a handset bought for home use can then access the two other application families. Indeed, in the telepoint service, such commonality will be needed to satisfy the Department of Trade and Industry's requirement that users should be able to roam between four service providers, thereby effectively mandating the introduction of a common air interface specification. This is now known as *CT2/CAI*.

This development has caused a reconsideration of the value of wireless access systems like CT2, and hence the expanded applications for this technology into, eventually, wireless-office systems. The radio link is therefore no longer seen as existing within a piece of telephone terminal equipment which is connected to a network via a conventional wired link, but is becoming itself a radio access port to networks.

The advantages of being able to purchase compatible handsets to any system from a number of suppliers will also produce a competitive supply situation, and allow flexible access arrangements to both public and private networks.

As a result, manufacturers and service operators from the UK have jointly produced a CT2/CAI specification[8], which is the subject of a companion paper[9].

CONCLUSION

The CT2 concept of a single-channel-per-carrier, time-division duplex, frequency-division multiple-access second-generation digital cordless telecommunications apparatus has been reviewed in respect of market requirements, spectrum availability, system architecture, and production implementation and specification. However, it is not sufficient to prove that CT2 satisfies the technicalities of the concept when its ultimate success depends on whether customers believe they are getting value for money. In order to minimise cost, therefore, the following points have been made:

- The core digital transmission hardware is suitable for the widest range of cordless telecommunication products thereby achieving economies of scale. In this regard, the ability to fully tap the potentially massive price-sensitive residential and small-business market is crucial.
- The basic system design has inherent cost-reducing features.
- The CT2 specification encourages innovative circuit design and production technology.

Finally, to ensure the widest possible success, the CT2 specification has been developed to identify a full common air interface specification. This encourages the creation of a standardised handset across the residential, business and public-access markets and effectively represents the first steps towards a universal personal communicator.

ACKNOWLEDGEMENT

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Biography

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CT2 Common Air Interface

M. W. EVANS†

Since 1982, cordless telephony has been available in the UK residential market using analogue CT1 products, such as the BT 'Freeway'. Recently, the Department of Trade and Industry (DTI) licensed four operators to provide telepoint services using digital CT2 technology, and soon CT2 products will also be available for the business market (wireless PBXs) and in the home. The fact that the customer will be able to use the same CT2 handset in these three applications has been made possible by the introduction of the CT2 common air interface (CAI) specification. This article describes the constituents of the CAI specification, and brings them together by describing the call set-up procedure from a CT2 handset to a base station (and vice versa) over the radio link.

INTRODUCTION

This edition of the *Journal* includes an overview article[1] describing the digital cordless telephony system known as *CT2*. This article goes on to describe further the protocol over the radio link (the air interface) from a cordless CT2 handset to its base unit and vice versa.

The initial CT2 specifications (BS 6833[2] and an associated radio specification MPT (Ministry of Posts and Telecommunications) 1334[3]) were written in the mid-1980s in an environment of liberalisation and minimum requirements to encourage innovation. The products then targeted by these specifications were cordless telephones with the same basic features as today's CT1 products (for example, BT's Freeway) and simple non-switched business products. However, it became evident with the advent of telepoint and wireless PBX applications[1] that there was a need for a handset user to roam between different telepoint operators' base stations and wireless PBX products. Therefore, the message protocols across the air interface needed to be well defined and common to all users. It was in this environment that the common air interface (CAI) concept for CT2 was developed.

At that time (early-1988), many companies, including BT, were investing heavily in CT2 product development, each with their own proprietary air interface. To bring about a common air interface standard, a committee was set up under the direction of the United Kingdom Department of Trade and Industry (DTI), and consisted of representatives from all CT2 equipment developers and the two major UK telephone network operators. The specification entitled 'Common Air Interface Specification to be used for the interworking of cordless telephone apparatus, May 1989', now issued as MPT 1375[4], is the result of many months of negotiation and detailed technical discussion by the committee members.

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COMMON AIR INTERFACE (CAI) DESCRIPTION

The common air interface specification can be broken down into five main parts:

- The radio interface
- Signalling layer 1
- Signalling layer 2
- Signalling layer 3
- Transmission and speech coding

Each of these parts is described more fully in the following paragraphs.

Radio Interface

Many radio parameters are specified in BS 6833 and MPT 1334 (for example, spectrum allocation of 864–868 MHz, 40 channels each of width 100 kHz, transmitted power of up 10 mW, intermodulation products, spurious radiation, transmitted power in adjacent channels, etc.), but the technique for modulating the radio carrier, and the time-division duplex rate (see below) were left to the discretion of the developer. The CAI now specifies the modulation to be two-level frequency shift keying (FSK) shaped by a Gaussian filter, with a frequency deviation of 14·4 kHz to 25·2 kHz above the carrier frequency (f_c) to represent the transmission of a binary 1, and 14·4 kHz to 25·2 kHz below the carrier frequency to represent transmission of a binary 0, as shown in Figure 1. This represents a modulation index of 0·4–0·7.

CT2 is a time-division duplex system in which the send and receive information between the handset and base unit (and vice versa) is transmitted in bursts interleaved on the same radio carrier frequency (f_c). The time-division

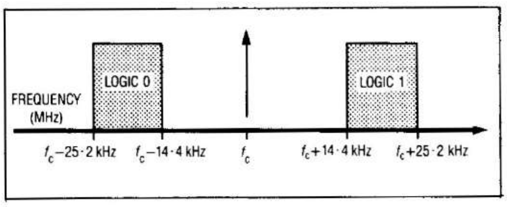


Figure 1—Modulation frequencies

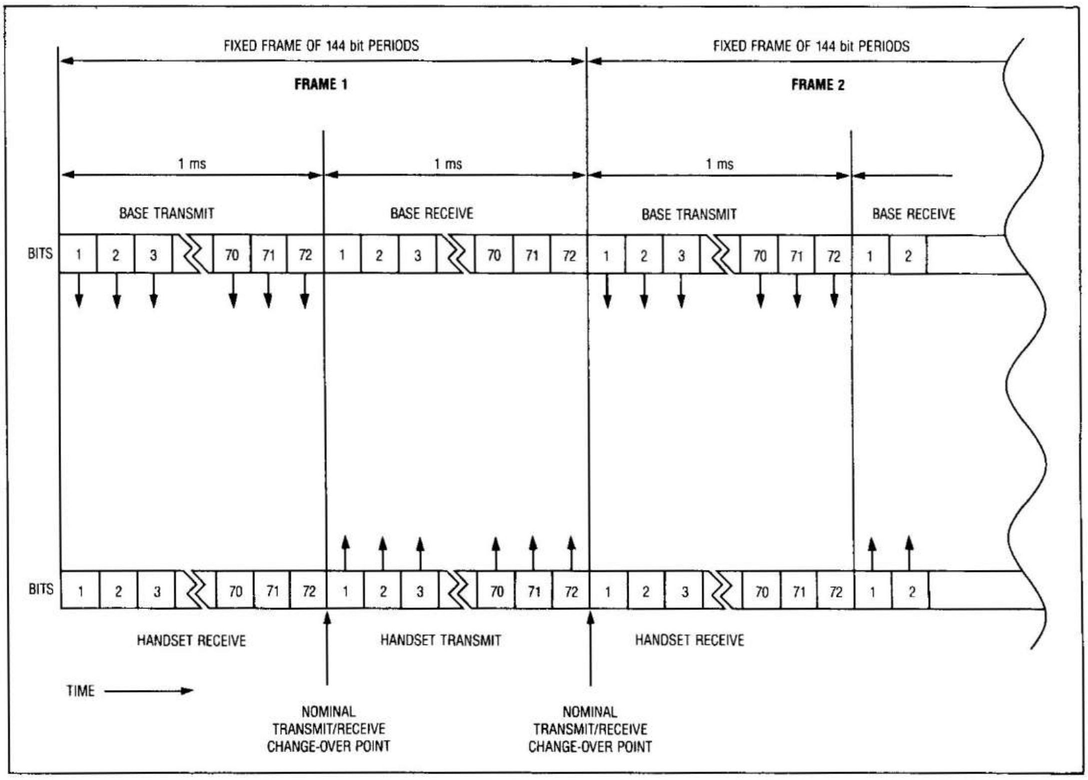


Figure 2
Frame format

duplex (sometimes known as the *ping-pong*) rate, that is, the rate at which base units and handsets switch between transmit and receive, is set at 500 Hz; 1 ms each for transmit and receive. Within this burst structure, there is a data rate of 72 kbit/s as shown by the frame format in Figure 2. In each burst, nominally 72 bits of data are available for speech, control, signalling and base–handset synchronisation purposes. However, there is an allowance for a dead time between bursts to allow the sending part to turn off its transmitter, and settle into receive mode, and for the receiving part to turn on its transmitter and settle at its centre frequency (f_C). This is nominally 4 bit periods long, but is shown in more detail later (in Figures 4 and 5). By use of this dead period, both ends of the link are sure that the receiver is able to decode accurately the first and subsequent transmitted bits in the burst.

In Figure 3, the transmission of a burst of information is shown. The amplitude of the radio signal is shown by the outer envelope, and the frequency (carrier and deviation) is shown for each bit of information. The crossing point between bits is at the carrier frequency f_C.

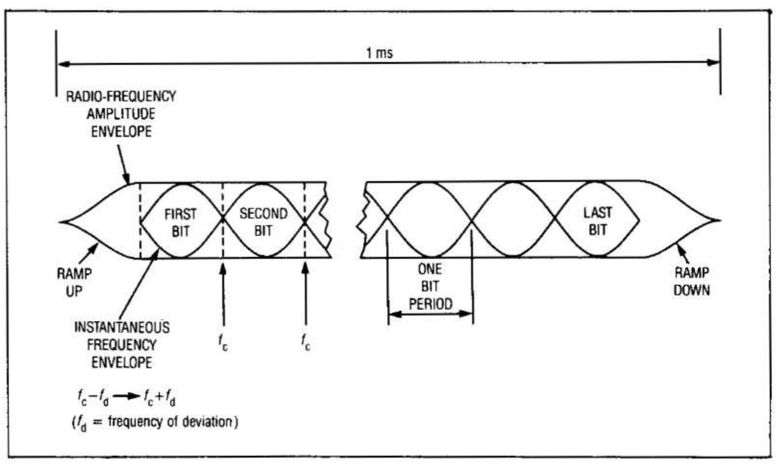
The radio interface is fully defined by BS 6833, MPT1334 and the additions described above. This radio interface has now to be used to carry speech, control, signalling and synchronisation information between a CT2 handset and base combination.

Signalling and Multiplexing

At the start of a telephone call, there is no synchronisation between the base and handset; in fact both units are scanning the 40 available radio channels for activity, or in IDLE mode (for battery saving in the handset), and are totally unaware of each other. In order to work together, they must first tune to the same selected radio channel, then attain bit synchronisation at 72 kbit/s, and finally burst synchronisation at the time-division duplex rate (500 Hz). Only after this is achieved, can meaningful information be passed over the link and decoded at the far end in order to set up a telephone call.

The various parts necessary for link set-up over the radio channel are constructed in a layered fashion similar to the first three layers of the Open Systems Interconnection (OSI). However, many parameters need to be defined in order to set up a synchronised radio link, and these are described in the following paragraphs. The whole process is brought together in the later sections of this article, by a description of the use of all these parameters in a call set-up from base to handset, and vice versa.

Figure 3
Transmitted burst showing a radio amplitude and frequency variation



Signalling Layer 1

Signalling layer 1 of the CT2 CAI specifies the method for two-way digital link initiation over the radio path and selection of the relevant information channel. It ensures that data and signalling channels can be established and maintained free from interference, where possible, over the air path. To do this, three information channels have to be multiplexed together within the instantaneous data rate of 72 kbit/s in various combinations depending on the requirements at the given phase of the call. These information channels comprise:

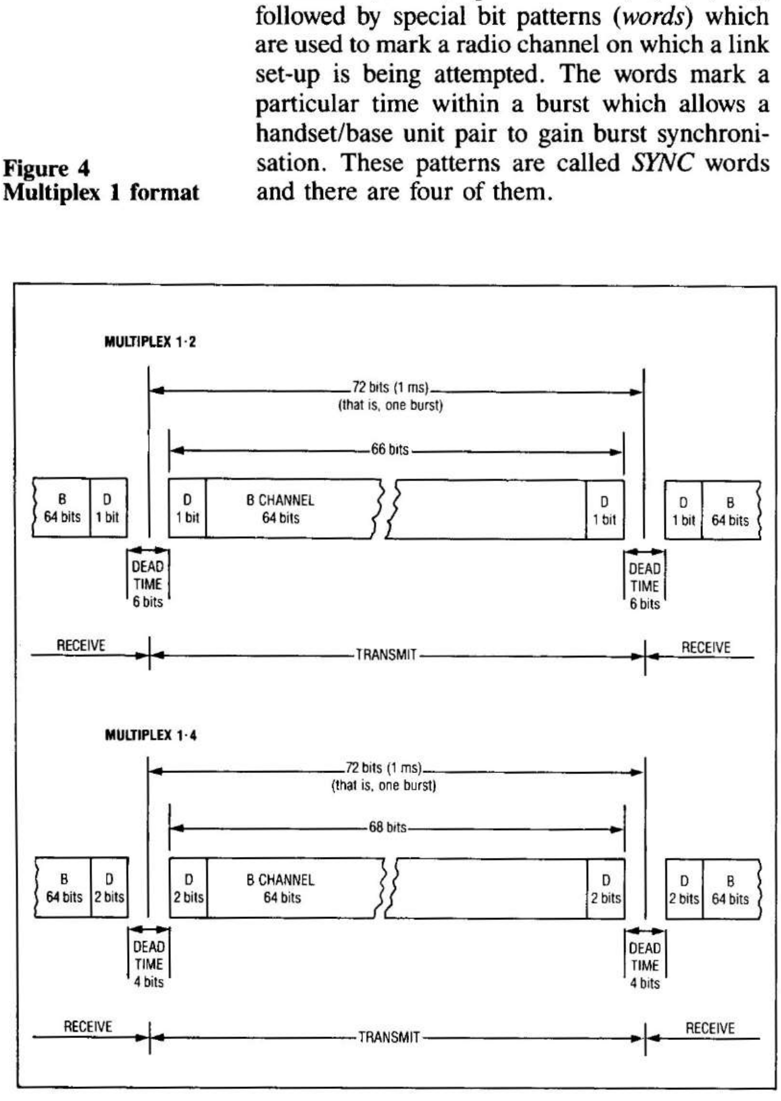
B channel (for speech or data),
D channel (for control and signalling), and
SYN channel (for bit and burst synchronisation).

An information channel may be absent in some circumstances, depending on whether the call is at the link set-up stage or in the speech state.

B and D channels are similar in purpose to those used in other communication applications, but the SYN channel is specific to CT2.

The SYN channel consists of a series of bit inversions (that is, a preamble of 1, 0, 1, 0, ...), followed by special bit patterns (*words*) which are used to mark a radio channel on which a link set-up is being attempted. The words mark a particular time within a burst which allows a handset/base unit pair to gain burst synchronisation. These patterns are called *SYNC* words and there are four of them.

Figure 4
Multiplex 1 format



Channel markers (CHM) are used when a base or a handset is attempting to initiate a call on a radio channel. The handset will send CHMP (P for portable) when initiating a call to a base, and the base will send CHMF (F for fixed) when initiating a call to a handset. When a link is already established, the handset will respond with SYNCP, and the base with SYNCF. The values of these words are shown in Table 1 and were chosen to yield low auto correlation and low cross correlation with other frequently occurring bit patterns. Each is 24 bits long. CHMP and CHMF are bitwise inverse, as are SYNCP and SYNCF.

The multiplexing of the B, D and SYN information channels gives rise to three separate multiplexes, known as *multiplex 1*, *2* and *3*.

TABLE 1
CHM and SYNC Words

CHMP	41B1AF hex	CHMF	BE4E50 hex
SYNCP	14E4FA hex	SYNCF	EB1B05 hex

Multiplex 1 (Figure 4)

Multiplex 1 is used when the handset and base unit are in the speech state. At this stage of the call, both ends of the link (that is, the handset and base unit) are in bit and burst synchronism, and so the SYN channel is absent. Speech is carried in the B channel, and the base and handset identities are interchanged via the D channel to assess the integrity of the link. If errors are detected in the received identity code sent over the D channel, the unit can consider whether the bit error rate is too high for acceptable speech quality, and attempt a radio channel change to a quieter channel while speech is in progress. This change happens rapidly, and will go undetected by the user. While D channel errors are corrected by retransmission, there is no error detection on the B channel.

Multiplex 1 is further divided into multiplex 1.2 and 1.4, according to whether there are two or four D channel bits available per burst. In multiplex 1.4, two of the bit periods used in the dead time of multiplex 1.2 are used to carry D channel information, giving multiplex 1.4 a total of 68 usable bits, and multiplex 1.2 a total of 66.

Theoretically, there is no reason to have multiplex 1.2, but this was one of the compromises made by the CAI Industry Committee, as some members found they needed this freedom in their design. Others preferred the faster D channel rate of multiplex 1.4.

In multiplex 1.2, the raw D channel data rate is 1 kbit/s, and in multiplex 1.4 it is 2 kbit/s. The error-corrected rates are about half of this figure. The B channel rate is 32 kbit/s; that is, 64 bits per one burst period of 2 ms.

The choice of whether multiplex 1.2 or multiplex 1.4 is used during the speech phase is

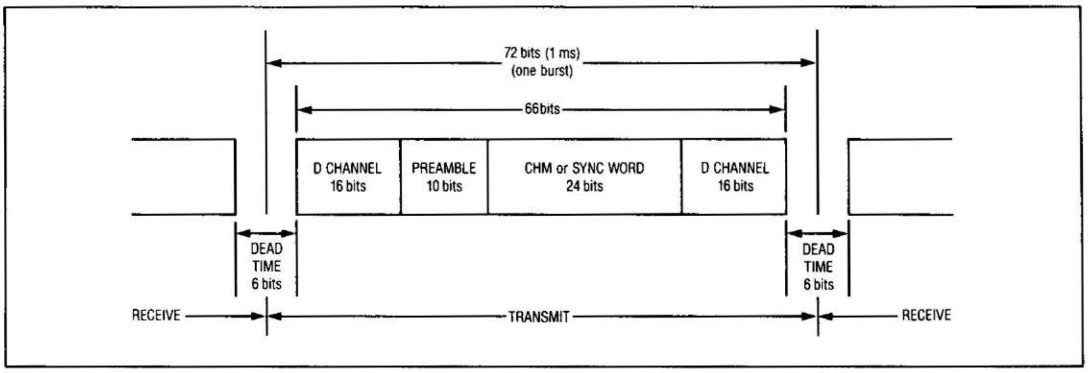


Figure 5
Multiplex 2 format

determined at the end of the link initiation phase by the interchange of messages in the D channel in multiplex 2.

Multiplex 2 (Figure 5)

Multiplex 2 is used during the link initiation phase of call set-up from the base unit when the base wishes to set up a radio link to its associated handset; for example, when an incoming call is received from the public switched telephone network (PSTN). At this stage, there is no speech; that is, no B channel. However, the base has to signal its identity to the target handset, and will use the D channel. It also needs to synchronise with the handset and uses the SYN channel to achieve this. Information is passed in a 66 bit format.

The SYN channel is made up of a 10 bit preamble and the 24 bit CHM or SYNC word. The preamble is a series of ten bit inversions (that is, 1, 0, 1, 0,...), which when detected by the handset enables it to gain bit synchronism with the base at 72 kbit/s. At this point, the handset does not know exactly where this bit pattern lies within the burst from the base. To achieve this, the CHMF word is transmitted directly after the preamble, commencing at a fixed point within the burst, and when detected allows the handset to synchronise its receive/send burst with that of the base.

During link set-up, the D channel carries link end-point identification information, which for the base is known as the *link identification code* (LID), and for the handset is known as the *portable identification code* (PID). The use of these codes is described during the link set-up procedure in the paragraphs on call set-up. In multiplex 2, the raw D channel rate is 16 kbit/s, with the error corrected rate of approximately half of this.

Having detected the base unit LID, the handset responds with SYNP in the SYN channel, and its own link identification; that is, the PID, in the D channel, in multiplex 2 format.

Multiplex 3 (Figure 6)

BS 6833 allows products to be developed that consist of more than a single handset/base unit combination. For example, it is possible to construct a multiple base with a number of transceivers serving a number of handsets. In this case, the multiple base transceivers should be synchronised so that they alternate between transmit and receive simultaneously. If this were

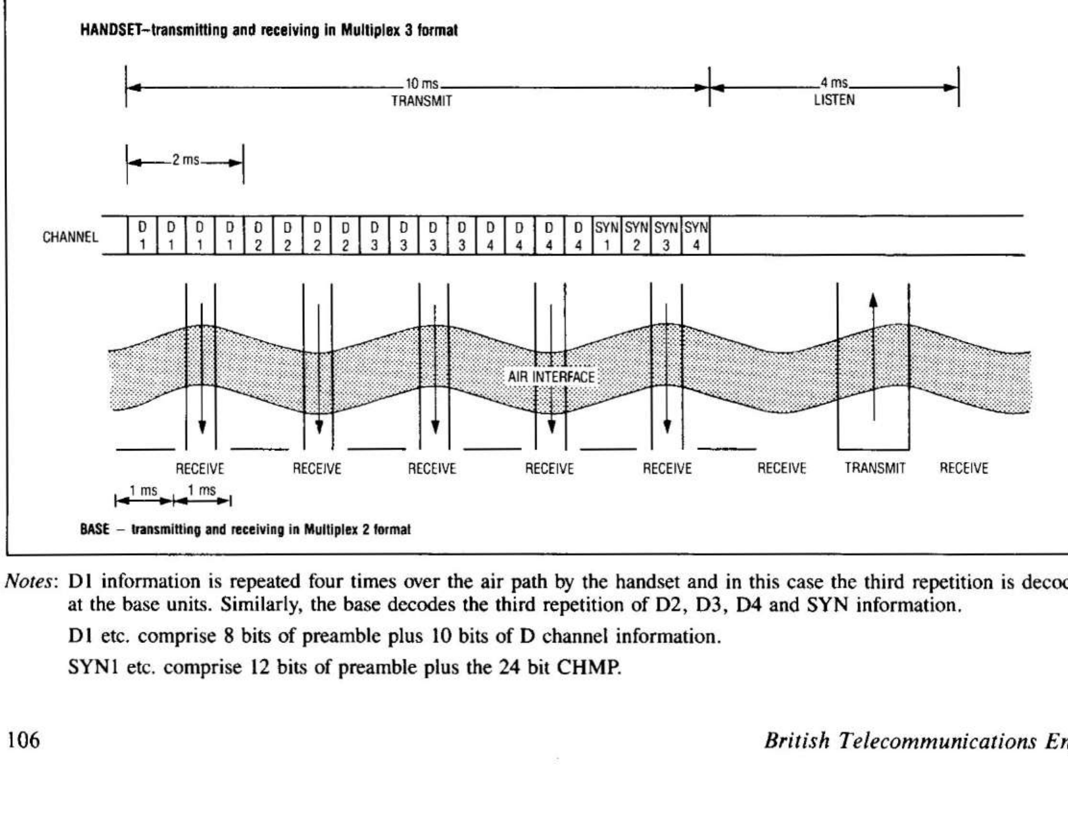


Figure 6
Multiplex 3 operation

Notes: D1 information is repeated four times over the air path by the handset and in this case the third repetition is decoded at the base units. Similarly, the base decodes the third repetition of D2, D3, D4 and SYN information.

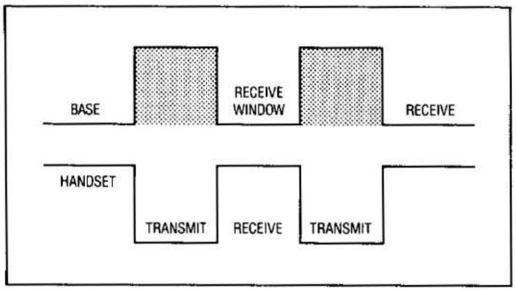
D1 etc. comprise 8 bits of preamble plus 10 bits of D channel information.

SYN1 etc. comprise 12 bits of preamble plus the 24 bit CHMP.

not so, it would be possible for one base transceiver in transmit mode to block (that is, overload) the input of another base transceiver in receive mode. For this reason, bit and burst synchronisation are determined by the base, with the handset acting as a slave off it.

If the handset wished to initiate a call and started to transmit and receive in multiplex 2 at the burst rate of 500 Hz, there is a possibility that the base and handset could be out of phase; that is, both in receive mode at the same time (see Figure 7), and the call attempt would be unsuccessful. This is not acceptable.

Figure 7



To overcome this problem, the handset attempts call initiation in multiplex 3, by paging the base for 10 ms with a relatively complex pattern of preamble (that is, 1, 0 inversions), D channel information (that is, its own PID and the target base LID), and the CHMP synchronising word in the SYN channel. Within the 10 ms signal, D and SYN channel information is repeated four times. After the 10 ms page, the handset sits in receive mode for 4 ms awaiting a response from its base unit. If none is received, it continues to transmit in multiplex 3 for up to 5 seconds.

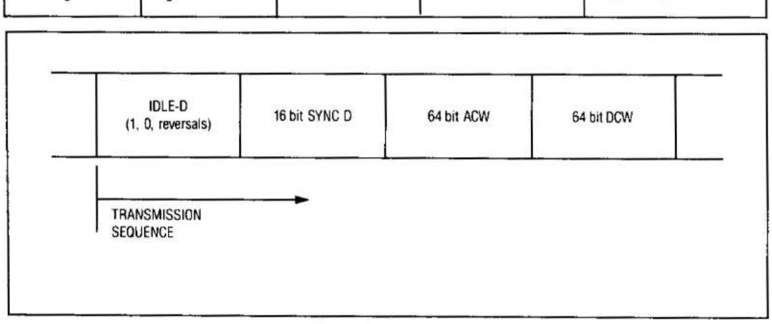
The base unit always operates at the 500 Hz burst rate, and therefore the repetition of information in multiplex 3 from the handset is designed to ensure one of each of the four repetitions of D and SYN channels is received at the base within the 1 ms base receive window as shown in Figure 6.

When the base has fully decoded the D and SYN channel, it transmits in multiplex 2 with its own identity (LID) and the handset identity (PID) in the D channel, and SYNCF in the SYN channel. The handset detects the SYNCF in its 4 ms multiplex 3 receive window and, knowing its exact position within a multiplex 2 burst, can then switch into multiplex 2 itself and be in synchronism with the base.

The uses of multiplexes 1, 2 and 3 are summarised in Table 2.

TABLE 2
Summary of the Uses of Multiplexes 1, 2 and 3

	Call Phase	Originating	Terminating	Channels Used D	B	SYN
Multiplex 3	Link set up	Handset	Base	✓	-	✓
Multiplex 2	Link set up	Base	Handset	✓	-	✓
Multiplex 1	Speech	either	either	✓	✓	-



Note: Assembled from multiplex 1 or 2

Figure 8—D channel structure

Signalling Layer 2

Signalling layer 1 provides the environment for a handset and base unit to interchange information in order to progress call set-up, call termination, etc. Signalling layer 2 covers the signalling channel protocols of error detection, error correction, message acknowledgement, link maintenance, and link end point identification. It also handles the transition from signalling layer 2 to 3. This signalling layer allows CT2 systems to communicate messages between end points, but does not define the meaning of these messages.

Layer 2 is based on the existing specification MPT 1317[5] which defines how packets of information are constructed to be sent over the radio link. An information packet consists of up to six code words, the first code word being the *address code word* (ACW), and subsequent ones being *data code words* (DCWs). Some messages can be sent within one ACW, in which case subsequent DCWs are unnecessary and therefore not used.

When there is no information to be sent over the D channel, a series of 1, 0 reversals (IDLE-D) is transmitted, and, prior to transmission of an information packet, a 16 bit synchronising word, SYNCD, is sent to alert the receiving end that a message is due for transmission. The format of the D channel is shown in Figure 8.

Both ACW and DCWs are made up of 64 bits in the form of eight 8 bit bytes. The first bit transmitted (that is, bit 1 of byte 1) determines if the code word is an ACW, when it is set to 1, or a DCW, when it is set to 0. The second bit of an ACW defines the format type (FT) of the message. The format is a fixed length for handshake or link end-point identification messages, when the FT bit is set to 0. With FT set to 1, the messages can be of variable length for transmission of information at layer 3 (see later section on signalling layer 3). FT is only significant in ACWs, and defaults to 1 in DCWs.

In general terms, of the 8 bytes within an ACW, bytes 1 and 2 are used for control, and bytes 7 and 8 for cyclic redundancy check and parity information. Bytes 3–6 are used to pass information. In a DCW, byte 2 is used for information rather than control, and this leads

to a useful data interchange rate of about half of the raw D channel rate.

Signalling Layer 3

A signalling layer 3 message is defined as a group of information elements delivered error free by signalling layer 2 to the far end of the radio link. In other words, it is at this level that meaningful telephony messages can be sent. Information elements can refer to the handset keypad, handset display, access to PBX facilities, and many more, and each element is broken down into further messages. For example, with the keypad information element set, the user has access to keyed digits 0–9, * and #and, with the display element set, messages can be sent from the base directly to drive a handset display. Layer 3 messages can be up to 29 octets in length, allowing product developers ample scope for all presently identified CT2 applications.

TRANSMISSION PLAN AND CODEC

In order to achieve the goal of the CAI, that is, universal interworking of handsets with base stations from different developers and offering differing services (for example, telepoint and wireless PBX), it is necessary to define the transmission characteristics of the handsets and bases, and to define the speech codec to be used.

The CAI defines a transmission plan which enables the transmission requirements of BS 6833 Part 2, 1987 (reference 2, which includes BS 6317 and BS 6305, references 6 and 7), and the ETSI draft recommendations for NET33, the European specification for interworking with digital networks, to be met. BS 6833 Part 2 defines the overall performance between the handset acoustic interface and a UK analogue telephone line, and NET 33 defines the requirements of the digital line interface. The CAI therefore acknowledges the emergence of the European transmission standards and provides an environment to meet them, but, in the analogue domain, alternative applicable standards may apply in different countries.

The preferred CAI codec algorithm is the CCITT Recommendation G.721 (1988); that is, the Blue Book definition of adaptive differential pulse code modulation (ADPCM). This is a complex algorithm, and, to reduce this complexity, and reduce battery power drain, a handset may be allowed to use a reduced specification ADPCM, but the overall speech path must meet the subjective and objective transmission tests specified in BS 6833 Part 2 (1987).

CALL SET UP

The preceding paragraphs of this article have described the elements required to set up a speech path over a CT2 common air interface radio link. The next two sections describe how the elements are combined to set up a link from base unit to handset, and vice versa.

Call Set Up from a Base Unit to a Portable Handset (Figure 9)

This call is initiated by incoming ringing signal from the PSTN being detected at the base unit. In response to this, the base scans the 40 available radio channels in turn in order to select a suitable one for the call. It either chooses one which has a noise level below a given threshold, which is defined as a maximum of 40 dB relative to 1 μV/m, or, where all channels have noise

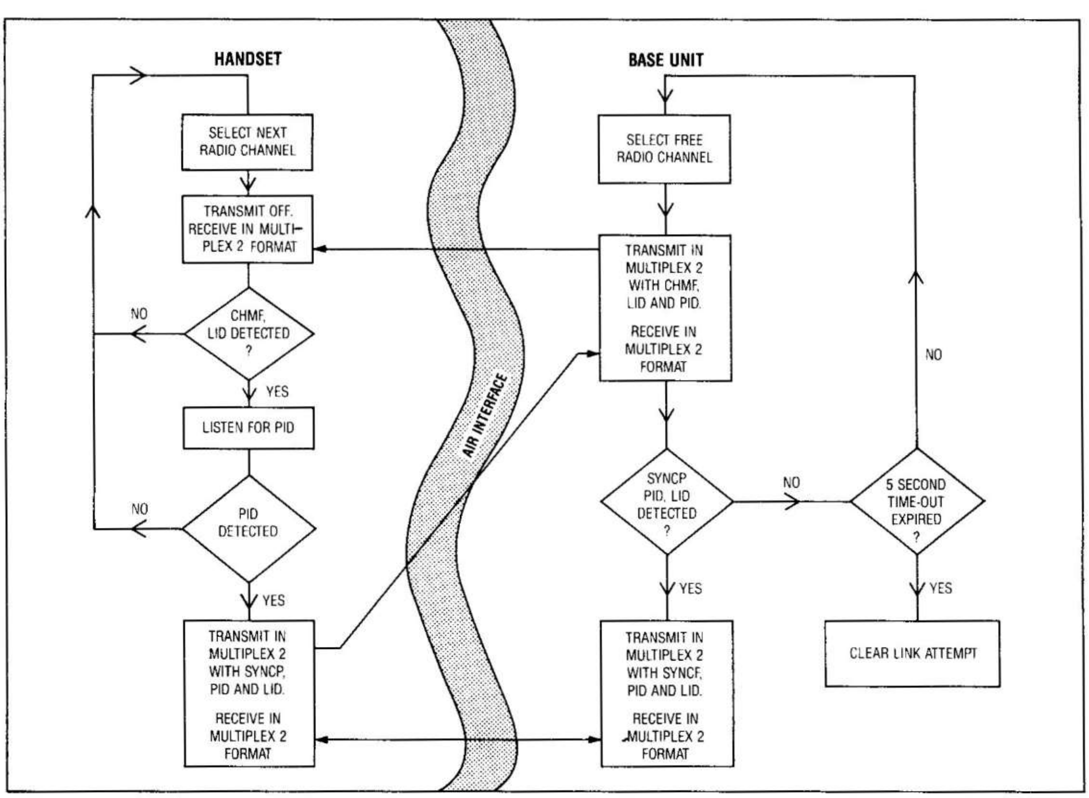


Figure 9
Link set-up from base unit to handset

above this level, it chooses the one with the lowest noise value. A channel that is being used for another telephone conversation is seen as having a very high level of noise, and will not therefore be selected.

This is the process of dynamic channel allocation, as the choice of radio channel is made at the beginning of each link set-up, and may be different from call to call.

The base unit then transmits on the selected frequency in multiplex 2 with its own identity (LID) and the target portable identity (PID) in the D channel, and the channel marker word for a base unit (CHMF) in the SYN channel.

At this stage, the handset is unaware of the incoming call, or the actions taken by the base unit. It could be in its SLEEP state, in order to preserve its battery charge, or radio-channel scanning. The ratio of sleep to scanning time could be of the order of 80%. In the SCANNING mode, the handset selects a start radio channel at random and looks for activity in multiplex 2 format. If none is detected, it moves onto the next. This is purely a listening function, and consumes a minimum of battery power.

When activity is detected on a radio channel, the handset remains on that frequency. It sets its automatic gain control (AGC) to a suitable level on receipt of digit information (the lower the radio signal, the greater the gain, and vice versa), and achieves bit synchronisation on the preamble of the D or SYN channels. It then looks for the CHMF word, and, if this is detected, it synchronises its burst to that of the base unit, as the position of the SYN channel within a burst is known. D channel information is then decoded and, if a valid LID is received, the handset remains on the frequency looking for its own PID. If a PID is received, but does not correspond to its own stored PID (that is the call is intended for another handset), the handset stops receiving on this frequency and moves onto the next. If it does receive its own PID, it then starts transmitting on the same frequency in multiplex 2 with SYNCP in the SYN channel, and its own PID and the base unit LID in the D channel.

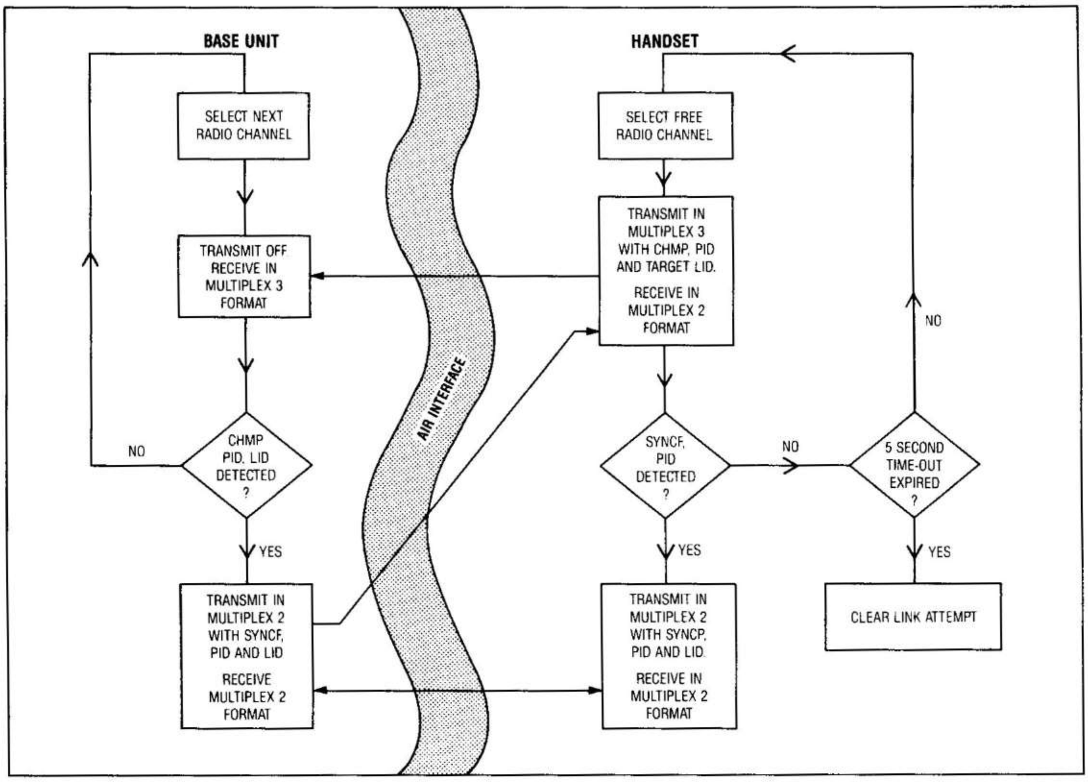
The base unit then detects the transmitted SYNCP, LID and PID, and informs the handset of this by transmitting SYNCF in the SYN channel instead of CHMF, continuing with LID and PID in the D channel. At this stage, a synchronised base-unit-to-handset link has been established.

If the base does not receive SYNCP, PID and LID within a given time, for example, because of impulsive noise on the radio channel, it selects another free channel. If it is not possible to set up a link within 5 s (usually allowing up to five attempts on different frequencies) the call would be aborted. However, as this is an incoming call from the PSTN, to avoid the caller receiving ring tone but no reply, each new cadence of ringing is interpreted as a new call.

When the link has been established, the base instructs the handset to turn on and off its calling device; for example, a tone caller.

If the handset user presses the LINE (or equivalent) button to answer the call, a layer 3 message transfer occurs to determine if multiplex 1.2 or 1.4 can be used in the speech state. This having been decided, the switch from multiplex 2 to multiplex 1 takes place and speech can commence over the B channel.

Figure 10
Link set-up from handset to base unit



Call Set Up from a Portable Handset to a Base Unit (Figure 10)

When a handset user wishes to make a call, he/she presses the CALL button on the handset. The unit then commences scanning the 40 available radio channels, selecting the first one with an interference level below the noise threshold. If none is available, it chooses the quietest one above the threshold. Again, this is dynamic channel allocation as discussed earlier. The handset then signals to the base unit in multiplex 3, with its own PID and the target base LID in the D channel, and the CHMP in the SYN channel.

At this stage, the base unit is unaware that the handset is calling, and is scanning each radio channel looking for activity. If none is detected on the current channel, it steps on to the next, and so on. If, however, incoming signals (that is, the selected centre frequency plus or minus the deviation frequency) are detected, it remains on that channel, firstly setting its AGC and then obtaining bit synchronisation from the preamble sequences in the D or SYN channels.

Figure 6 shows that one SYN channel submultiplex, and therefore one CHMP word, can be decoded in each 10 ms of multiplex 3 transmission. On receipt of this word, the base then goes on to decode the D channel information comprising the base unit LID and the portable PID. If the LID does not match with its own stored value, the base returns to channel scanning. However, if the LID does match, the PID is decoded and the base then transmits in multiplex 2, with SYNCF in the SYN channel, and the received PID and LID in the D channel.

If the PID and LID are not received from the handset within a defined time bit period, the base unit continues radio-channel scanning by selecting a new radio channel.

The handset, on receipt of SYNCF, PID and LID, ceases transmission in multiplex 3, and moves into multiplex 2, synchronisation with the base being achieved by the known position of SYNCF in the burst. It informs the base of successful receipt of PID and LID by sending SYNCP to the base in the SYN channel instead of CHMP.

A synchronous link has now been established and the base unit can seize a line to the local exchange or PBX/key system. In order to return dial tone to the user, the system must move into multiplex 1 as this tone is transmitted over the B channel. The base and handset therefore interchange messages to decide on whether to use multiplex 1.2 or 1.4, and, after switching to multiplex 1, dial tone is received over the air path, enabling the user to input keypad information. Dialled digit information is passed over the link as layer 3 messages.

Call Collision

The preceding paragraphs describe the call set-up sequence under normal conditions. However, under certain circumstances, it is possible to get call collision. For instance, a handset may be calling a base on one radio channel, and the base may be calling the handset on another. In this situation, no responses are possible from the other end of the link, and a stalemate persists until one end clears down. This will most likely be the handset, as it will cease calling after 5 s if it cannot establish a link to its base. In this case, the incoming call can then be detected by the handset.

Another instance when call collision can occur is in the wireless PBX situation, where several bases are situated within a building, each capable of connecting the portable CT2 handset to a PSTN line, another PBX extension or a private circuit. When a handset sends out a *calling* signal (that is, in multiplex 3), it is possible to get responses from many base stations within range. However, the chance of detecting simultaneous multiple responses is low, as all bases start their channel scanning at a random channel number, and therefore there is a random time at which the handsets call is detected at each of the base stations. The first to detect the calling signal handles the call.

However, if a collision does occur, that is, two or more base stations respond to the handset on the same radio channel at the same time, the situation is resolved as follows. The handset always calls in multiplex 3 for a set period, and, if it cannot decode the response it receives from the base stations, it selects a new channel and again calls in multiplex 3. The base units will be waiting for the handset to respond on the initial frequency in multiplex 2 with the SYNCP word in the SYN channel. If this is not received, they each restart channel scanning at a random channel number looking for handset calling activity. In this way, the first base to settle on the new handset frequency takes the call.

CONCLUSION

A common air interface for CT2 digital cordless telephony has been described, listing the major parameters required to set up a radio channel link between a portable handset and base unit, and vice versa. This was followed by a description of call set-up using these parameters with calls being originated from either the base unit or handset.

All CT2 products developed for approval in the UK must now conform to the CAI specification, and this will ensure interworking of handsets on simple cordless products, telepoint systems and wireless PABXs supplied by different developers and manufacturers.

Recently, a Memorandum of Understanding was signed by a number of European Community members agreeing to adopt the CT2 CAI specification as a basis for telepoint systems in their countries. This will allow handset users to move between these countries and make calls

via the different telepoint systems using only the one CT2 handset.

ACKNOWLEDGEMENTS

The author would like to thank colleagues in BT for all their help and perseverance in writing the CT2 common air interface specification, and the other members of the industry group for their positive approach to writing the specification and their expert technical input. The companies involved are Ferranti Creditphone Limited, Libera Developments Limited, GEC Plessey Telecommunications Limited, Mercury Communications Limited, Orbitel Mobile Communications Limited, Shaye Communications Limited and STC plc.

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Biography

Martin Evans obtained B.Sc. and Ph.D. degrees in Physics at University College, London, in 1966 and 1972, respectively. He joined the Post Office in 1972 and has worked on telephone and local line transmission, on the evaluation of ITT proprietary PABXs (the Pentomat and Unimat range), and on the development and support of the Monarch range of PABXs. More recently, he has been responsible for managing the development of CT2 products for BT and their application in wireless PBXs, telepoint, and domestic cordless environments.

The Cashless Services System

N. G. POPE†

In November 1988, BT introduced an automatic version of the Telephone Credit Card Service. This new service called BT Chargecard *still allows calls via the operator, but also allows customers to make calls from any public telephone or other telephone with multi-frequency (MF4) capability and to transfer automatically the charges to their home or office telephone bill. The equipment used to introduce this new service forms a kind of intelligent network. This article describes the equipment used, how the development was done and the way to the future for telecrediting.*

HISTORY

For many years up to the end of 1988, BT operated a little known service called the *Telephone Credit Card*. This service allowed a customer to make a call via the operator and to transfer the call charges to his/her home or office telephone account. Customers were issued with a card similar to the one shown in

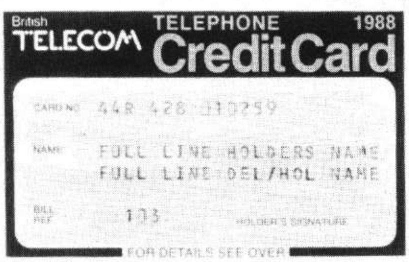


Figure 1—Telephone credit card

Figure 1. This card had a number printed on it which the customer quoted to the operator. After a check of the number, the operator would put through the call, and the call charges would appear on the telephone bill associated with the card number. Although the service was never promoted by BT, the numbers of card holders gradually increased to about 250 000. The card could also be used in over 120 countries around the world, but calls were restricted to those made back to the UK only.

Telephone credit cards are issued by many countries, but their usage was restricted to calls via the operator until the mid-1980s when card issuers in, particularly, the USA began to issue automatic credit cards.

An automatic telephone credit card usually retains all of the features of an operator card and calls can still be made via the operator. However, an automatic card allows the call to be set up without the intervention of an operator. The process is quicker and easier for the customer and is cheaper for the telephone company.

† National Payphone Services, British Telecom UK

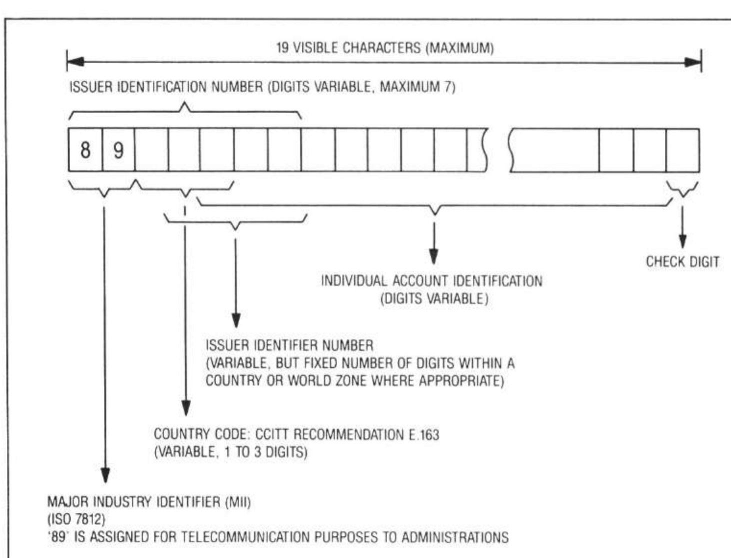
In 1985, BT decided to begin the process of introducing an automatic card in the UK. This began with a trial of the service in the Bristol area starting in 1986 and ended with the introduction of the service in November 1988. During the trial, the service was called *AccountCall*. This name was found to be unsatisfactory as it caused customer confusion, and the service is now called *BT Chargecard* (BTCC).

Customers have been very pleased with the new service which has seen rapid growth since the automatic system was introduced. This growth is expected to continue at a high rate over the next 3–5 years.

CCITT RECOMMENDATIONS

Operator-controlled telephone credit card operation is regulated by CCITT Recommendation E.116. This is, in fact, a simple document laying down a few rules about card size, the numbering scheme and the method of operation. During the last study period, it was recognised that this Recommendation was inadequate for the more exact specification needs of an automatic operation. A new Recommendation E.118 was developed specifically to deal with automatic cards

Figure 2
E.118 numbering system



and this appeared in the CCITT Blue Book at the end of the study period.

Recommendation E.118 is a much more specific document. It states that cards must conform to International Standards Organisation (ISO) standards and, most importantly, it introduces a new international numbering system suitable for world-wide automatic operation.

Figure 2 shows the new numbering system. The number consists of a prefix which identifies the country of origin and the card issuer plus an account number which is unique to each customer. BT's card numbers have the following form:

89 44 00 DD Y NNNNN

where:

89 is the major industry identifier for telephone cards,

44 is the country code for the UK,

00 indicates a BT-issued card,

DD is the BT District code,

Y is the year digit (controls reissue of cards), and

NNNNN is the customer's unique account number.

The CCITT has also introduced a further Recommendation (E.113) which is a communication system to allow the use of cards in other countries. More details of this are given below.

THE BRISTOL TRIAL

Before committing itself to the expense of developing the equipment for a national automatic service, National Payphone Services decided to run a trial to establish customer reaction. In 1986, equipment obtained from Plessey (now GPT) was installed in the Bristol and Bath group switching centres (GSCs). An ICL System 25 computer in Bristol was also installed to form a simple database for management of customers' accounts.

The exchange equipment was installed in the grading in such a way that it was able to provide service to most payphones in the centres of Bristol and Bath. Customers were recruited using various trial marketing techniques.

The trial results were very encouraging, and much useful data was obtained on customers' needs and preferences. This was then input to the technical specification for the national system.

THE BT CHARGECARD

Figure 3 shows the BT Chargecard. The card has the same shape, feel and construction as any other credit card, and is essentially a reminder to the customer of the account number and service operating instructions. Possession of the card is not essential to make a call either on the manual or automatic service.

The card features two numbers, and the customer is also provided with a secret personal identification number (PIN) for use with the

Figure 3—BT Chargecard

automatic service. The upper number in large characters is the customer's account number for use with the automatic service. The smaller, lower number is the account number used via the operator and when the customer is calling from outside the UK. The numbers are essentially the same, but the operator number has a CCITT prefix and two check digits on the end.

Automatic calls are set up in the following way:

Customer dials the access code—144.

A voice message says 'Please dial your account number and PIN'.

Customer dials 12 digits on keypad.

The system validates the account number/PIN pair.

A voice message says 'Please dial the telephone number you require'.

Customer enters the destination number.

Voice guidance (in five languages) is available throughout call set up. Messages may be interrupted or pre-empted by practised users.

Three levels of service are offered to customers:

- *Full service* allows calls to any destination in the world.
- *National service* allows inland calls only.
- *Specified number only service* allows calls only to one number specified in advance.

Customers with full and national service also have a call home number which can be accessed by a short code.

Daily credit limits are available to customers to control expenditure, and all calls are fully itemised on the home or office telephone bill.

THE CASHLESS SERVICES SYSTEM

There are various ways in which an automatic Chargecard service can be introduced into the network. Ideally the service would be built into the software of an intelligent local exchange with account number/PIN validation being carried out in an intelligent network database (INDB). This is still considered to be the ultimate solution to the problem of providing the service. However, in 1986, this solution was not a practical proposition and so an alternative had to be found. After considering many different possibilities, the cashless services system was devised.

The cashless services system (Figure 4) is essentially an overlay intelligent network. It comprises two new items of equipment:

(*a*) The cashless services processing unit (CSPU). These items are located at 30 digital main switching unit (DMSU) sites and provide the exchange functions of the system.

(*b*) The cashless services database (CSDB). A central intelligent database used to keep records of customer accounts and to validate account numbers and PINs before a call is allowed.

The name 'cashless services' was chosen as it was recognised that this system has the potential to provide many different intelligent network services which require high-speed validation and special billing.

A call is set up in the following way (refer to Figure 4).

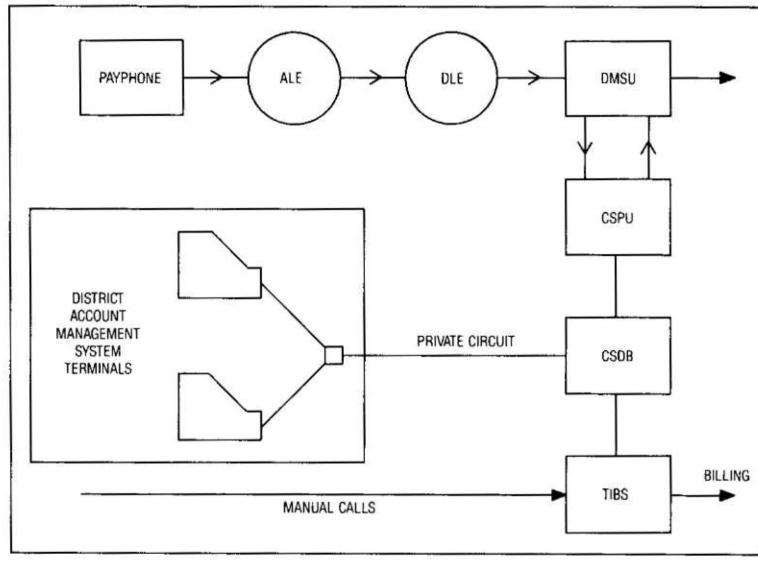
The call in Figure 4 is shown originating from a payphone, but in fact any MF4 capable telephone can be used.

When a customer dials 144, the call is routed to the nearest digital node. (A digital local exchange is shown in Figure 4 but this is not the only possible routing.) This digital node adds three digits identifying the originating charge group before passing the call on to the DMSU and ultimately to the CSPU.

The call terminates on the CSPU and is effectively a free call. There now exists a through transmission path between the customer and the CSPU. The CSPU can communicate with the customer by using voice messages and the customer can communicate with the CSPU using MF tones. Note that payphones connected to exchanges which do not have MF capability signal 144 by using loop-disconnect and then automatically switch to MF for all subsequent digits. Dual-signalling telephones may have to be switched to the MF mode by the customer at this point, usually by dialling *.

When a call terminates on the CSPU, the CSPU sends out a one-second burst of 1600 Hz tone. If the originating telephone is a payphone, this triggers the payphone to send forward its identity to the CSPU using MF tones (9 digits). Ordinary telephones do not respond to the tone. The tone is followed by a message from the CSPU's voice guidance system which asks the customer to dial his/her account number and PIN. The CSPU then receives MF digits from the customer and stores them. Voice messages are repeated if the caller does not respond and may be changed to other languages by the caller at any point during the call set up. Language selection codes are:

*1	English (default language)
*2	Welsh
*3	French
*4	German
*5	Spanish



ALE: Analogue local exchange
DMSU: Digital main switching unit
CSDB: Cashless services database
DLE: Digital local exchange
CSPU: Cashless services processing unit
TIBS: Telephone input billing system

Figure 4
Cashless services system

When the CSPU has received the 12 digits representing the account number and PIN, it must validate these numbers before allowing the call set up to continue. In order to do this the CSPU sends a message over an X.25 communications link to the cashless services database (CSDB). The CSDB is a four-processor non-stop intelligent database. The incoming PIN is encrypted and compared with the encrypted PIN stored against the account number given. As these validations are against the customer's actual records as stored in the database, this system is known as *positive validation*. It is the most secure method of guarding against fraud.

Validation takes about 100 ms and when it is completed the CSDB sends back a message to the CSPU telling it to allow the call and giving details of the customer's account. The CSPU then sends a further voice message to the customer asking him/her to dial the telephone number required. If the validation should fail, then the CSDB sends back a different message instructing the CSPU to give the customer an appropriate message explaining what is wrong.

The CSPU receives and stores further digits from the customer until sufficient have been received to allow the call to be routed. The call is completed via the DMSU and the public switched telephone network (PSTN). The CSPU remains in the circuit throughout the call and is the supervisory point for cashless calls.

At the end of the call, the CSPU sends a further message to the CSDB. This is the billing message and contains all of the details necessary to construct an itemised billing record; for example, start time, duration, originating charge group, destination number etc. The CSDB then performs a billing calculation and allocates the appropriate call charge before passing the call record into the existing BT billing system via the telephone input billing system (TIBS) computer in the normal way. Records of manual calls made

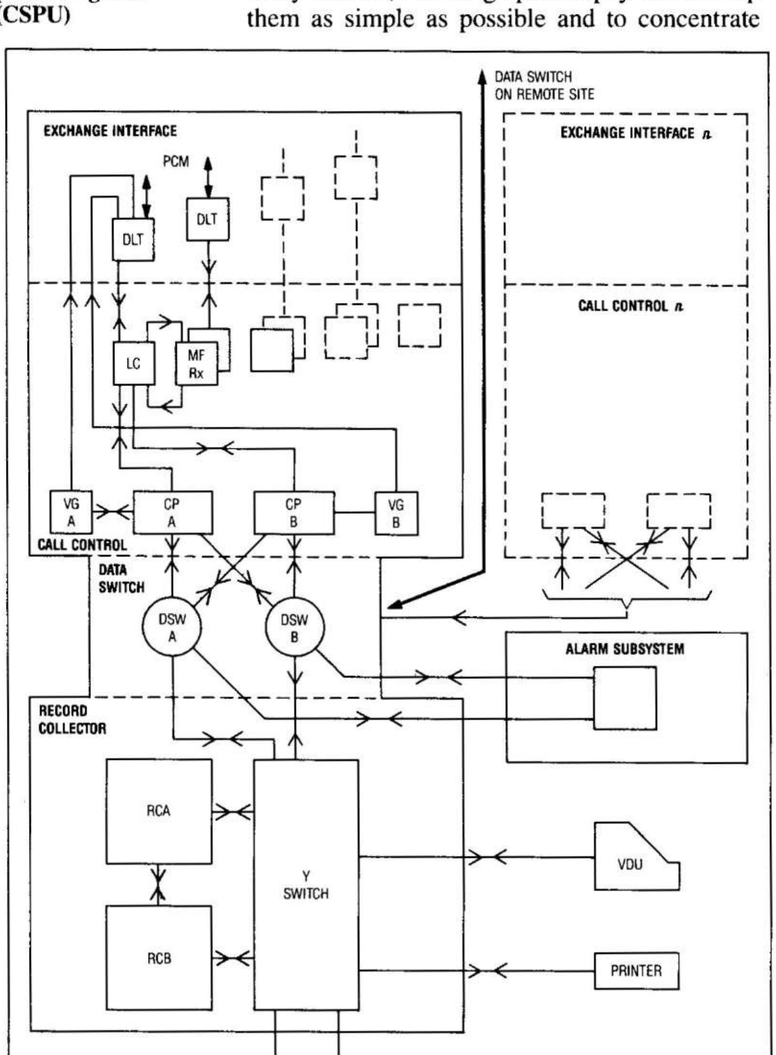
via the operator are also fed into the billing system via TIBS from the appropriate manual board system; for example, ACRE, OSS.

In order to process calls, the CSDB must contain records of all customers having Chargecards. This data is entered via the account management system (AMS). Each District has a Chargecard office with one or more AMS terminals. These terminals are CSS compatible for ease of operation and allow Districts to maintain records of all customers in a paperless form. New customers may be added, details changed or enquiries made, all in real time via the AMS.

Cashless Services Processing Unit (CSPU)

The CSPU performs the exchange functions of the cashless services system. There are 30 CSPUs scattered throughout the country to minimise routings for cashless calls. As there are so many CSPUs, the design philosophy was to keep them as simple as possible and to concentrate all of the intelligence into the CSDB so as to make changes and upgrades as easy and cost effective as possible. However, there must be some intelligence and sophistication in the CSPU as it has to set up and monitor the call on its own with very little assistance from the CSDB. The CSPU was developed to a BT specification by GPT in Liverpool. A block diagram of the CSPU is shown in Figure 5.

Figure 5
Cashless services processing unit (CSPU)



PCM: Pulse code modulation
LC: Line controller
CP: Call processor
RC: Record collector
DLT: Digital line termination
VG: Voice guidance
DSW: Data switch

The CSPU is an all-digital device. Calls are routed into and out of the CSPU over 30 channel PCM systems using TS16 signalling. Calls are not decoded to analogue at any time. Each incoming channel has a corresponding outgoing channel, so no switching capability is required. The CSPU can be divided into five main parts (see Figure 5). These are the exchange interface, the call control, the data switch, the alarm subsystem and the record collector.

(*a*) The exchange interface handles the incoming and outgoing PCM interface with the DMSU. Data is converted from HDB3 into binary PCM for processing by the rest of the CSPU. Clock recovery and PCM alignment are also carried out at this point.

(*b*) The call control monitors the incoming lines for TS16 signals and for MF digits. The MF receivers are specially designed for this application. They have increased sensitivity to cope with the transmission levels which can be expected on some analogue 144 routes. The call control also provides the voice guidance to the customer via duplicated voice guidance systems. The voice messages are real voices, digitally encoded and stored in EPROMS. This gives a much more user friendly feeling to the system than speech synthesis. The call control is the supervisory point for all calls, monitoring them and timing them. Any failures are reported to the alarm subsystem.

(*c*) The data switch is the CSPU's internal communication system. It allows the flow of messages between all parts of the system and is duplicated for security.

(*d*) The record collector consists of a pair of DEC Microvax computers running in hot standby mode. It provides system operators with facilities for control and management of the CSPU. It also provides the X.25 signalling facilities between the CSPU and the CSDB.

(*e*) The alarm subsystem monitors faults and failures from all parts of the CSPU. Alarm conditions are signalled to maintenance staff via the DMSU alarm system. Alarm messages are also sent to the CSDB for remote monitoring of the system as a whole.

Each CSPU can handle up to 720 erlangs of traffic and is designed to grow as traffic demand increases. Call controls and data switches may be provided on remote sites so as to make routing of traffic on the 144 network easier and to reduce the overall system costs. In the current cashless services system, there are 10 central CSPUs with record collectors. The remaining CSPUs are on remote sites.

Cashless Services Database (CSDB)

The CSDB consists of a four-processor Tandem VLX computer. It can be seen that if the CSDB should fail then the whole of the cashless services system will cease to function. This may appear to be a major risk, but the VLX computer is designed specifically for this type of non-stop operation. All hardware and software within the machine are duplicated and the system is capable of fault-tolerant continuous operation. It was a difficult decision to design the system with only one database, but experience of the first year of operation has shown the decision to be correct.

Figure 6 shows a block diagram of the CSDB indicating the subsystems.

The card management subsystem is the most complex. It interacts with terminals in each BT District to allow customer details to be reviewed, added or changed on-line. Changes entered by the Districts go through a checking process before being added to the main customer and accounts file. Since interaction is directly with this main file, customers can be given quick and accurate responses to enquiries. When a new account is added, details are sent to the card and PIN producing companies on magnetic tape. A card and PIN are then produced and sent to the customer by post.

The validation subsystem has the task of checking account numbers and PINs before calls are allowed to proceed. This process must take place as quickly as possible to ensure that the customer is not kept waiting. Validation messages are sent from the CSPU via the X.25 network, and the incoming account numbers and PINS are checked against records in the validation file (a subset of the main file). Currently validation takes about 100 ms to complete. This activity takes priority over other CSDB functions to ensure the best possible performance from the system.

The pricing subsystem deals with the billing messages sent from the CSPU at the end of the call. The call details are extracted and the call is priced in accordance with the current pricing tables held in the CSDB. An itemised record is then assembled and is sent to the appropriate billing system on magnetic tape. All calls are archived on optical disc. Current Chargecard tariffs are similar to those charged for ordinary calls and details of the tariff tables are obtained from the national charging database (NCDB). However, the CSDB is capable of very flexible call pricing and a wide variety of tariff structures could be applied if required.

The CSDB has an extensive range of statistics and management information. Reports are available as print outs and on-line.

Fraud could be a problem on this type of service unless it is kept strictly under control. The fraud management subsystem constantly monitors fraud levels and activities taking appropriate action as required.

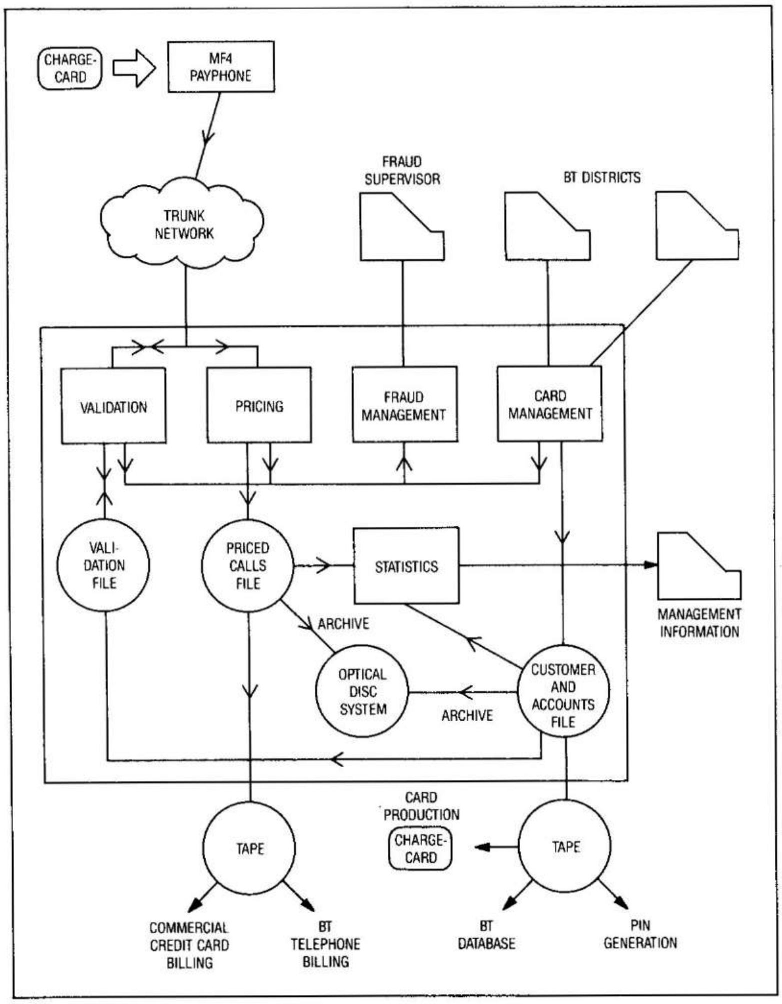


Figure 6
Cashless services database (CSDB)

CREDITCALL

BT has over 500 payphones capable of accepting payment by commercial credit card (Figure 7). These are mainly provided at major travel and tourist sites throughout the country and they are able to read data directly from the magnetic stripe of a credit card. When this service was first introduced, the credit cards were validated in a dedicated central computer and it was necessary to run a separate pair of wires from each telephone back to the central point. This was a very expensive process. The introduction of the cashless services system has allowed a much simpler and easier way of providing the service.

Now, when a customer makes a call from a CreditCall telephone, the telephone reads the card number and then automatically dials 144. When the CSPU responds, the telephone sends the credit card number to the CSPU using MF signalling. The card number is then passed on to the CSDB for validation. This now means that the extra private circuits are not needed and CreditCall telephones can be provided at any location in the country. CreditCall telephones accept major credit cards and the BT Chargecard can also be read by these telephones which saves the customer the trouble of keying in the account number.

Figure 7
CreditCall telephone (Cardphone 2A)

THE FUTURE

The cashless services system adds intelligence to the network and is capable of providing other services if required. Of particular interest is the flexible off-line billing system which is capable of producing itemised billing records with a wide variety of tariff structures. Other services are currently being considered, but they will have to form part of a future article.

However, one particularly interesting development which can be discussed now is the introduction of international credit card validation. At present, over 120 countries accept the BT Chargecard, but only for calls back to the UK. This is mainly to restrict fraudulent use overseas. It is planned to interconnect the CSDB with other equivalent databases abroad. This will allow a BT card to be used in another country but still be validated against the record in the CSDB. This development will allow the card to be used for domestic and third country calls when overseas and will remove the restrictions on calls to the UK only. It is planned to have this service working in its initial form in 1992.

In the longer term, as more digital exchanges are introduced into the network, it is planned to move the CSPU functions into the local exchange. Such a move will require system modifications in local exchanges but will make more efficient use of the network. At the same time the CSDB will become part of the larger INDB.

For the short term, a major expansion of the Chargecard user base is under way, and Districts are having to work flat out to keep up with the demand. A bright future lies ahead for cashless services.

Biography

Nigel Pope's first job was safety equipment maintenance for the National Coal Board. In 1974, he obtained an honours degree in Electronic Engineering from Liverpool University and then joined BT via open competition. He has been involved with many kinds of system development especially associated with the TXE4 exchange. In 1984, he moved into signalling and was involved in interworking testing on CCITT No. 7 equipment. In 1986, he joined National Payphone Services to lead the team responsible for the development of the cashless services system. He has now moved on to become Cashless Services Manager responsible for product management of Phonecard, Chargecard, Creditcall and mobile payphones.

Recent Developments in Silicon Design: A BT Viewpoint

A. B. M. ELLIOT†

This article describes recent advances in silicon technology, the methods used for integrated circuit design and the supporting computer-aided design (CAD) tools. Examples of customised integrated circuits designed at the British Telecom Research Laboratories are used to illustrate the vital role played by silicon design technology in the development of new telecommunications systems and services.

INTRODUCTION

Developments in silicon integrated circuit (IC) technology have been at the heart of many of the major developments in telecommunications and information technology in the past three decades. Without large-scale integration of circuits on silicon, digital switching and transmission would be only an interesting concept, personal mobile communications would be a pipe-dream and the office automation revolution could never have happened.

Standard IC components, such as microprocessors and memories, have been an important element in these developments, but custom-designed silicon is of increasing importance and, consequently, it is the fastest growing sector of the market. Custom silicon enables companies to differentiate their products from those offered by competitors and it enables system developers to achieve results that are only possible with very high levels of integration. Many other advantages can flow from the use of custom silicon, including lower cost, size, weight and power, and higher performance and reliability. Custom silicon comes in three main guises: gate array, full custom and standard cell. All three design styles are supported at British Telecom Research Laboratories because each method has its unique advantages; more details about each technique are given in later Sections of this article.

Silicon designers in BT have been involved in many of the important technological advances that were predicated on IC developments. Included among the important contributions made by BT designers are the development of high-reliability, high-performance regenerators for submerged and land-line repeater systems, one of the first programmable microprocessors, a world-beating 9·6 kbit/s modem chip set and various application-specific integrated circuits for System X and the Monarch PABX.

This article describes some of the recent developments in silicon technology, design methods and computer-automated design (CAD) tools, and explains how BT designers have used these commercial developments to develop the new chips and chip sets used in various new BT systems and products.

SILICON PROCESS TECHNOLOGIES

The phenomenal increases in integrated circuit density continue at an almost unabated pace. Although there has been some reduction in the logarithmic rate of increase, complexity is still doubling every 18–24 months. Memories and other circuits which contain large arrays of identical blocks can quickly take advantage of the improvements in process technology, but less-regular very large scale integration (VLSI) circuits are now limited by the complexity of the design process rather than the capability of the silicon processing. Today's design tools and methods have problems in managing circuits with over a million transistors.

This Section examines the trends in the complementary metal-oxide semiconductor (CMOS) and bipolar processes used for gate array, full custom and standard cell designs. There have also been important developments in mixed bipolar-CMOS (BiCMOS) technologies but these are not examined in any detail. The BiCMOS developments tend to have taken two separate and distinct routes. One set of processes has been optimised for high-speed digital applications, where the bipolar transistors are used to increase the drive to heavily loaded internal nodes and external pins. This enables complex systems integrated on a single chip to operate close to the intrinsic speed capability of the CMOS transistors: 200 Mbit/s system operation should be possible. The other set of processes is targeted at mixed analogue/digital applications, often operating at higher voltages and taking advantage of the larger dynamic range of the bipolar transistors for analogue operation.

CMOS Technology

Silicon suppliers often use the same process for different custom design methods, or indeed for standard parts. In fact, many manufacturers are

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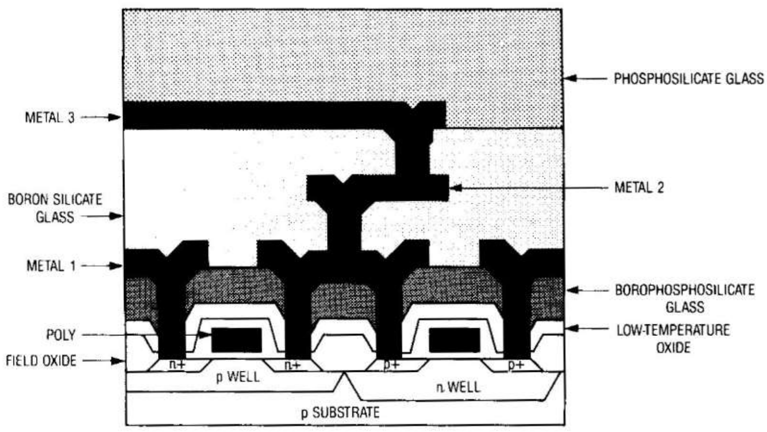
now settling on a unified set of design rules to enable blocks of merchant parts to be used as macros in standard cell designs.

Most standard production processes use a gate length of 1 μm or larger but some manufacturers are introducing sub-micron processes for gate array designs. These sub-micron processes may soon be available to BT designers for full custom designs through strategic partnerships with major silicon suppliers. They are made available on the understanding that the process may be subject to small changes before reaching production status.

Figure 1 shows a typical process used for a 1 μm sea-of-gates gate array. This process uses three layers of metal to get a gate utilisation factor of 75%; many standard cell and full custom processes would be similar. Availability of the standard cell option in a given process may lag by about a year behind the gate array because of the time required to design and characterise all the cells.

Figure 1 Cross-section of a typical 1 μm CMOS process for a sea-of-gates gate array



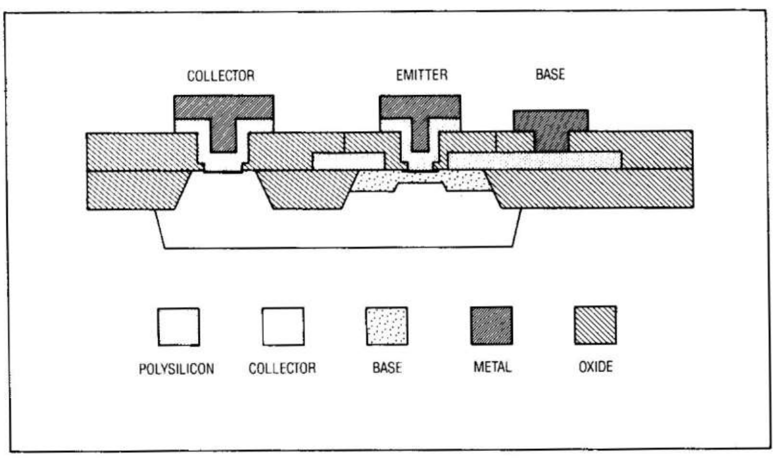
Bipolar

The same bipolar processes are also often used for the various design styles, but there is very limited availability of standard cell libraries for bipolar technologies.

An example of an advanced high-speed bipolar production process is the MOSAIC III process from Motorola Semiconductors. This process is used by Motorola for standard parts and the MCA 3 gate arrays, and British Telecom Research Laboratories (BTRL) has negotiated foundry-level access for full custom digital and analogue/digital designs. The process uses a self-aligned polysilicon contact structure to produce transistors with effective emitter widths of less than 1 μm and small intrinsic device parasitics from photolithographic line widths of 1·5 μm. The unity gain bandwidth (F_t) is greater than 15 GHz and emitter-coupled logic (ECL) gate delays of less than 100 ps have been demonstrated.

The cross-section of the Mosaic III transistor shown in Figure 2 illustrates how the double-polysilicon process is used in a unique way for the self alignment of the emitter and base contacts. The double-polysilicon process also enhances the analogue capability of the technology by providing additional polysilicon components such as resistors, diodes and capacitors. Side wall oxidation of the first polysilicon layer is used to produce the sub-micron emitter width from standard photolithographic techniques. The circuit metallisation uses three layers of metal interconnect with polyimide as the interlayer dielectric (but the two upper layers have been omitted from Figure 2 to simplify the diagram). The final passivation also uses polyimide but this is capped with silicon nitride to exclude any surface water vapour.

Figure 2 Cross-section of a Motorola MOSAIC III high-speed bipolar process transistor



DESIGN STYLES AND DESIGN FLOW

The three main categories of custom integrated circuit are as follows:

(*a*) The *gate array* design style, in which a common set of uncommitted circuit elements is built into the silicon substrate and the wafers can be held as stock with the first layer metal interconnect deposited but not yet etched. A specific circuit option is configured by the final layers of interconnect.

(*b*) *Standard cell* chips are designed by building the circuit from design cells held in software libraries supplied by the vendor. CAD tools are used to place the cells and route the interconnect. All layers of the circuit are unique for a given circuit option and the method requires a full set of photographic masks for each circuit type.

(*c*) In the *full custom* design style, the circuit is constructed from the transistor level up, without the use of pre-designed circuit blocks.

Although new design methods are being introduced that enable these methods to be mixed on a single silicon die, this is still a useful starting point for discussing developments in design styles.

Gate Array

Early gate arrays had the cells of gates arranged in regular columns or rows with channels between these for routing the interconnections. This arrangement can be used for circuits with

up to about 20K gates. Recently, gate arrays have been introduced which employ a so-called *sea-of-gates* architecture, where the routing is not confined to channels but runs over the gate structures. This allows the gates to be packed much tighter and, although gate utilisation is not as high as with channelled arrays, over 75% utilisation can be achieved using multi-layer metal technology with three layers of metal. Circuit densities can now be over 100K usable gates on a single chip.

An example of the sea-of-gates technology used in recent BTRL designs is the 1 μm HDCMOS gate arrays from Motorola (see Figure 3).

Another development in gate arrays is the inclusion of blocks of high-density RAM, but the problem with this approach is that the blocks have to be of fixed size and therefore are seldom a good match to the requirements of a specific design. An alternative method of implementing RAM is to compile it from the basic gate array structure. Although this approach is more flexible, it consumes more silicon area, using 1·5–2 gates per bit.

Standard Cell

Early standard cell designs used fairly simple cells, equivalent to TTL SSI functions, designed as *fixed height* blocks. These were assembled in fixed height rows or columns and channel routed in much the same way as gate arrays. As the complexity of the cells grew, it was increasingly difficult to constrain every cell to a fixed height, so random geometry cells were introduced and more sophisticated routers developed. Complex fixed geometry cells were not always the answer: for circuit blocks such as RAM, ROM and programmed logic array (PLA), each circuit needed a different size and organisation of the block. To solve this problem, compilation tools were developed that automatically constructed the layout for each block as required for a specific circuit.

A modern state-of-the-art standard cell system may include:

(*a*) multipliers, accumulators, barrel shifters;
(*b*) register stacks;
(*c*) various analogue blocks, including 12 bit analogue-to-digital and digital-to-analogue;
(*d*) microprocessor and microcontroller cores;
(*e*) microprocessor peripherals;
(*f*) bit slice microprocessors; and
(*g*) data-path compilers for RAM, ROM and PLA.

Full Custom

At a high conceptual level, the design methods for full custom and standard cell can be considered similar. A full custom design starts with a floor-plan of the chip that divides the circuit into the major blocks. Each block is then broken down into smaller blocks and the process repeated until the blocks are sufficiently small for them to be built up from the transistor level. The main difference between this and a standard cell design is that, in the standard cell method, detailed knowledge on silicon layout is not necessary because someone has already done some of the work by pre-designing the blocks. The advantage of the full custom approach is clearly that it gives the designer more flexibility and the disadvantage is that it can take more time and therefore cost more.

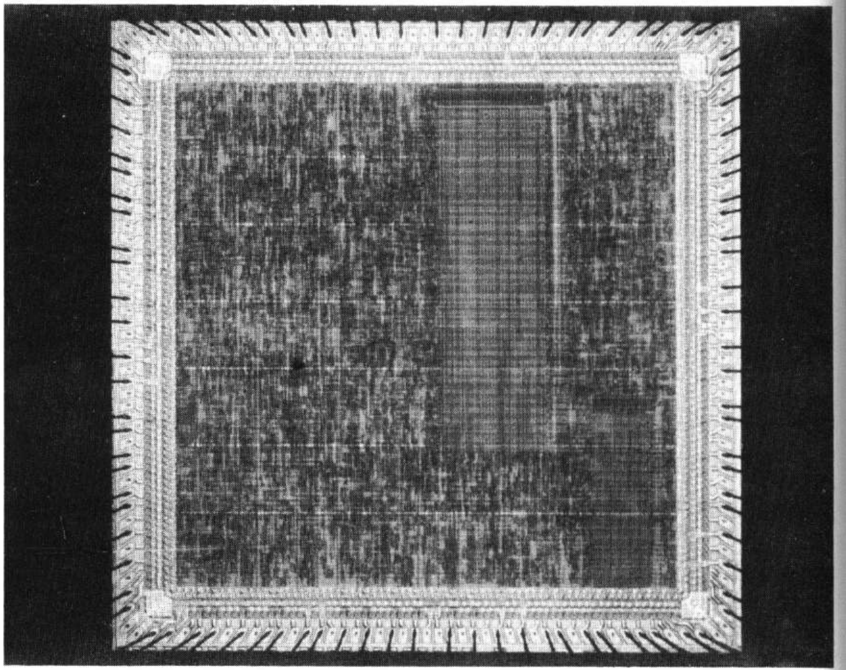


Figure 3
Finite impulse response filter chip

Full custom methods are appropriate for high volume, low cost and lower volume, high performance designs.

Design Flow

In general terms, the design flow is similar for all design styles (see Figure 4).

The major classes of CAD tool required to support this design flow are:

- behavioural simulator
- schematic capture
- logic simulator
- fault simulator/test generator
- layout tools, including place and route.

In full custom designs, further tools are needed to support the circuit level design processes that are transparent to the end user in the other two design styles.

CAD TOOLS FOR DESIGN, MANUFACTURE AND TEST

Gate Array

Many low-cost PC-based design systems have been released in recent years and, although these systems have a role to play, any potential user should be aware of their limitations. They can

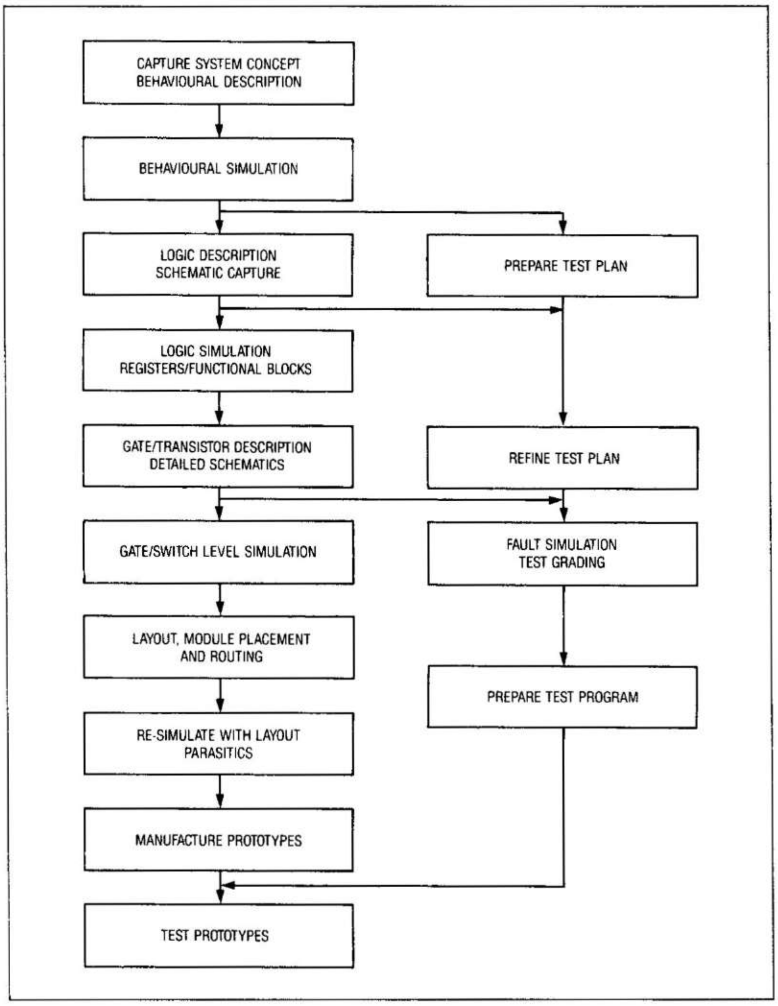


Figure 4
Generic design flowgate array, standard cell and full custom designs

be used for schematic capture and circuit simulation of circuits with a few hundred gates, but the design-proving and test-generation procedures for large circuits, which today may have more than 100K gates, need powerful high-MIP (million instructions per second) workstations with substantial amounts of system memory.

Traditional design tools experience difficulties in handling the high complexity sea-of-gates structures now being used; new approaches will be necessary.

One problem relates to the traditional approach to resolving timing problems revealed by post-layout simulation. Prior to layout, the estimated delay is derived from the fan-out of each gate and a statistical average value for the metal interconnection. After layout, some of the nodes will have higher than average interconnection length, and the gate may need to be replaced with a higher drive version and the automatic place-and-route tools run again. For large gate arrays, the track loading can be so dominant in a large sea-of-gates circuit that each re-run generates new problem paths. New smarter place-and-route tools are required to overcome the problem. One approach is to constrain the placement of gates within a given functional block to the same physical area of the gate array. Gate delay estimates can then be based on the size of the functional block for all gates internal to the block; only gates driving other blocks would have their delay estimates based on the larger interconnect length.

Faster clock rates and larger arrays present special problems with clock skew that can be further exacerbated if level-sensitive latches are used with their dual-phase non-overlapping clocks. One solution to this is incorporated into the Tangate place-and-route software from Cadence; it employs a clock tree synthesis option that tackles the problem by partitioning the clock net according to the physical placement of circuit functions (that is, gates are clocked from local buffers). The associated routing algorithm equalises the track length between the source buffer and the gate sinks; this contrasts with conventional routers which try to minimise track length.

As far as testing is concerned, the mechanistic use of traditional techniques, such as scan-path testing can result in unacceptable silicon overheads. A richer set of 'design for test' methods is required and more intelligence in the tools or in their use.

Standard Cell

The previous section showed that the emergence of 1 μm and sub-micron technologies has made possible the integration of 16/32 bit processors, compiled RAM/ROM and other generic LSI functions on a single chip. The economic justification for this level of integration is not always self-evident; component costs can be higher than using standard parts and the justification usually relies on reduced system costs (board space, power etc.) or specific size and power constraints.

The economic advantages can disappear if adequate design systems are not available, particularly for those components that are not required in millions. Complex designs need good system (behavioural) level simulators to ensure that optimum architectural decisions are made before logic design starts. These tools are only beginning to appear and, although library support from application-specific integrated circuit (ASIC) vendors is limited at present, it is developing fast, and routes from the hardware description languages (HDLs) used in these simulators, through logic synthesis tools such as the design compiler from Synopsys Inc., to semiconductor vendor gate array and standard cell libraries now exist.

Additional development problems arise with those IC designs that use processor cells because of the need to verify not only the system hardware, but also the software running in real time, before committing the chip design to silicon. One approach is to use core elements bonded out as separate chips together with re-programmable logic cell arrays to implement the remaining ASIC net list, (a flexible system of this sort is commercially available).

The emergence of direct-write electron beam technology has reduced the cost of prototyping and low-volume production. Instead of using photographic masks to define circuit geometries, they are defined by a electron beam focused onto the wafer surface. The electron beam is controlled by a computer in the e-beam machine and it is easy to 'write' different circuits for different customers on a single wafer. Although the technique can be applied to gate array, it has a more dramatic effect on the cost of standard cell parts, because they would otherwise need a full set of masks for each design option. This development could have a radical effect on the economics of IC production. Given good CAD support tools, prototyping in silicon becomes a realistic option and, for very complex chips, it may be cheaper to verify designs in silicon, than running extensive simulations on expensive high-performance computers.

Full Custom—The ASTRA Design System

BTRL uses the ASTRA design system, a full custom design tool that uses a symbolic representation of the circuit elements (see Figure 5). This system, which was developed at BTRL, has the advantage that the target process does not need to be defined until late in the design cycle.

The ASTRA system is based on the use of two distinct forms of symbolic design data to describe the design: hierarchical floor plans and cell layouts. From this design data, both the information required to check and simulate the design and the geometrical mask data used to fabricate the chip are automatically generated.

The design of the chip starts with some specification of the required behaviour or function of the chip. Using the design method supported by ASTRA, the first stage in the design process is to define the basic architecture of the chip and its associated floor plan. This involves partitioning the chip into an interconnected set of blocks, each with a defined function and an estimated rectangular size. The partitioning is done by a designer based on his/her expertise and experience, and is entered into the design system by using the graphical floor plan editor.

Once a floor plan has been entered, it can be used to generate input data automatically for simulation programs. The structural information it contains can be combined with suitable functional specifications of the blocks and relevant waveform data.

After the top-level floor plan has been captured, partitioning can be applied in a similar way to the main functional blocks, again using the graphical editor. This partitioning can proceed down the hierarchy until the resulting functional blocks are sufficiently simple to be implemented directly or until a block is specified for which a design exists. At each stage of the partitioning process, simulation and testability analysis can be applied to verify the quality of the design, and the resulting input and output data can be accumulated for later use in testing.

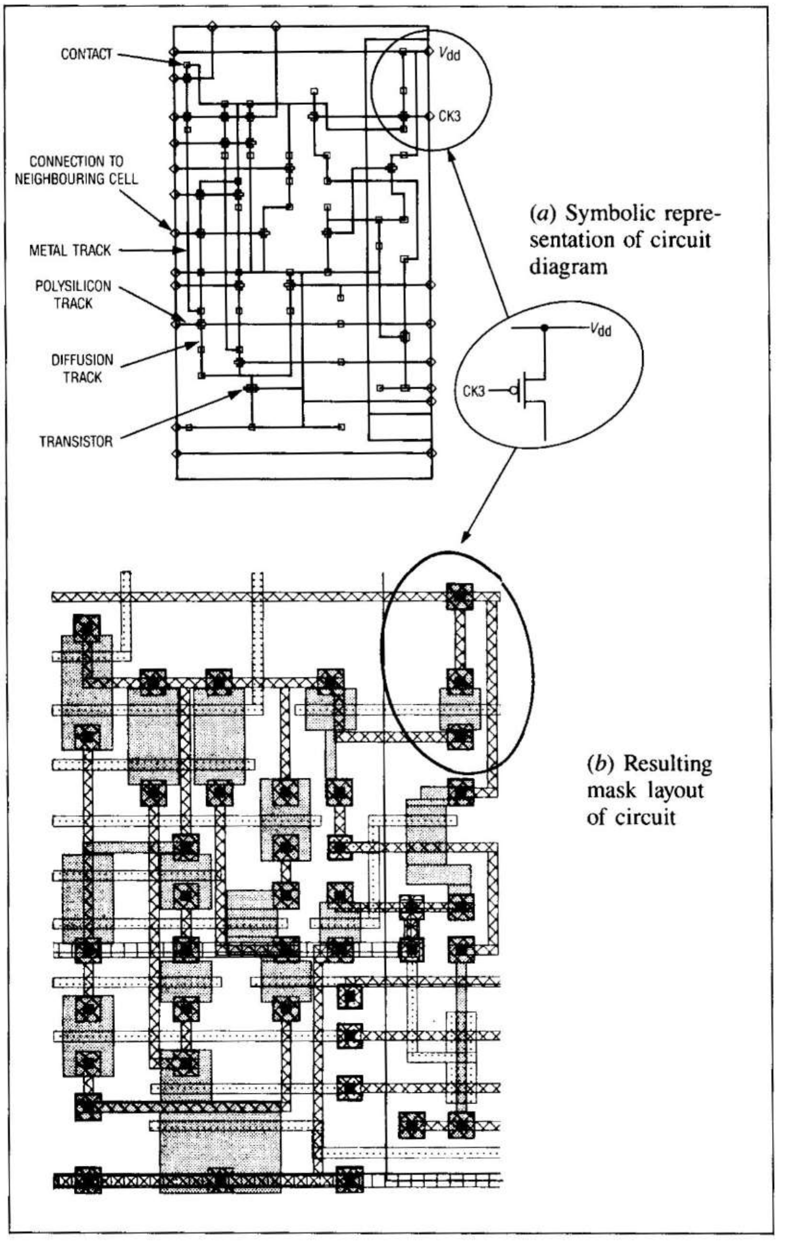


Figure 5
ASTRA defined symbolic and physical layout

The lowest level blocks or cells are defined symbolically using the leaf cell editor (Figure 5 (*a*)). This is used to create, move, modify or delete circuit elements such as transistors, contacts and lengths of interconnect of specified material.

When partitioning is complete, and all cells have been designed, the complete chip is assembled using the chip assembler program. For each of the constituent cells, the spacing program will have produced a file defining the minimum size and associated pin positions of each cell. The assembler works through the hierarchy defined by the floor plan, stretching cells and blocks as it goes to provide pitch matching between pins. No provision is made for 'spaghetti' interconnections, as all connections between blocks are made by abutment. However, blocks containing only routing are supported and can be created automatically using a switch box router.

At the end of the chip assembly process, the exact size and pin positions of all constituent cells are known. The spacer has produced mask geometry data (Figure 5 (*b*)) which, when combined with the hierarchical information extracted from floor plans, forms the mask geometry data files for post processing. These are processed in the conventional way to produce the masks.

RECENT PRODUCT DESIGNS

In this section, some recent chip developments in support of BT-designed products are described. The examples selected illustrate the use of gate array designs with an external supplier and the use of full custom CMOS in a collaborative activity with Mitel Semiconductors.

Video Codec Chip Set—A Gate Array Design

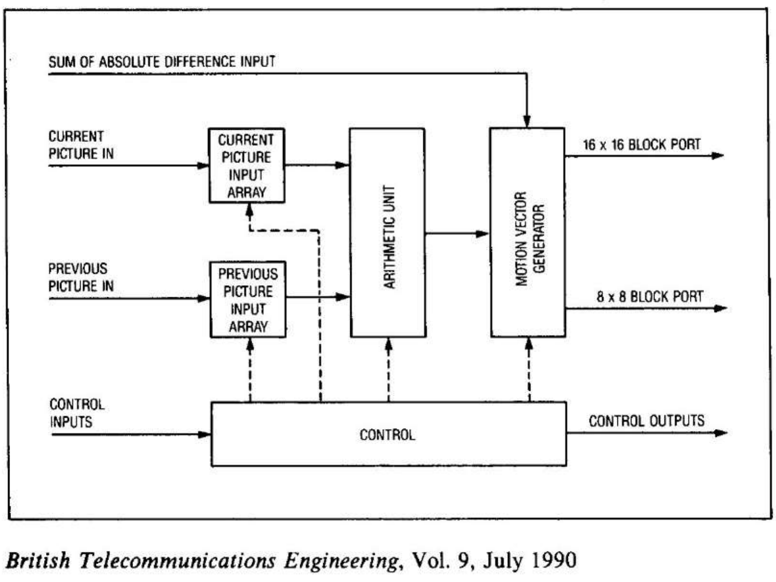
The semi-custom IC design group at BTRL designed a chip set for the Victel video codec during 1989. The chip set contained three circuits which performed the following functions within the codec:

(*a*) motion vector estimation,
(*b*) finite impulse response (FIR) filter, and
(*c*) control.

Initial studies of the motion vector estimation and filter circuits revealed that the gate counts for the two circuits were in the region of 48 000 gates and 20 000 gates, respectively. The circuits were to be designed to very tight timescales that precluded the use of standard cell technology. It was necessary to use the smallest geometry gate array devices in order that the high gate counts required could be realised at minimum cost.

The HDCMOS technology from Motorola was selected because this 1 μm sea-of-gates technology had triple-level metal interconnect and was capable of achieving high utilisation of the available gates: the use of triple-level metal increased the maximum utilisation to 70–80% compared with about 50% for double-layer metal. This had a substantial impact on the cost of the motion estimation circuit in particular as the design could be implemented using a 64 000 gate array rather than a 100 000 gate array. At this size of array, prices rise very rapidly with size of array as yields of good die per wafer are quite low.

Figure 6
Victel motion vector estimation circuit



(*a*) *Motion Vector Estimation* The function performed by the motion vector estimation circuit is to determine the minimum sum of differences between groups of pixels in successive picture frames. Eight motion vector estimation circuits are required to perform motion estimation on a 16 × 16 pixel block at a pixel rate of 6·75 MHz. The bulk of the circuit comprises an input register array followed by an arithmetic block (see Figure 6). The circuit is totally synchronous and was implemented in 45K gates of a 64K gate die (utilisation = 70%).

(*b*) *FIR Filter* The second circuit designed was required to implement a finite impulse response (FIR) filter. The circuit contained five RAM blocks to hold coefficients, five 8 × 8 multipliers and several adders. This circuit was implemented in 22K gates of a 31K array (utilisation = 71%).

The RAM areas are readily identified in the chip photograph shown in Figure 3.

(*c*) *Control Circuit* The control functions required for the eight motion estimation circuits together with the re-ordering of the pixel data were also implemented using gate array technology. Although there were only 5K gates of logic, a 16K gate array had to be used to obtain the required number of inputs and outputs.

Simulation

The motion vector estimation circuit was the largest gate array that had been designed at BTRL; though the design itself was relatively straightforward, the size of the circuit stretched the capabilities of the CAD software. Simulation times were lengthy, despite the relatively short test sequences required to verify the design. The size of the data structures caused the Apollo/Mentor design systems to page fault which heavily extended the simulation times further. The problem was alleviated by replacing the gate-level models for the adders in the circuit with a description written in Pascal. The effect was to reduce the size of the data structures and the simulation durations to the order of 4–5 hours. The final simulations prior to design release had to be performed with the vendor's gate level models for the adders. These simulations were performed on an Apollo system with 12 Mbyte of memory and took approximately 12–15 hours to run. New design methods will be required for larger circuits.

Layout

In the earlier section on CAD tools, reference was made to the post-layout timing problems

that can arise because of the large variations in the metal interconnect, and these were experienced for the two larger arrays in this chip set.

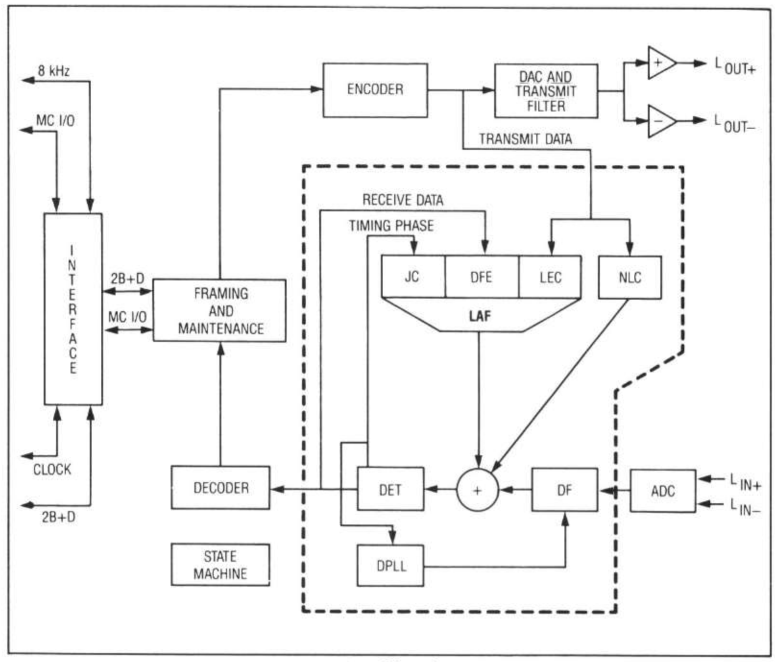
The observed variations in timing delays were much greater than had been experienced in previous designs and a greater number of gates required changing to high-powered variants. In the case of the motion vector estimation circuit, six layout iterations were required before the design was acceptable. As each layout together with data transmission time and checking time required one week, a large number of layout iterations is not desirable.

2B1Q Transceiver—A Full Custom Design

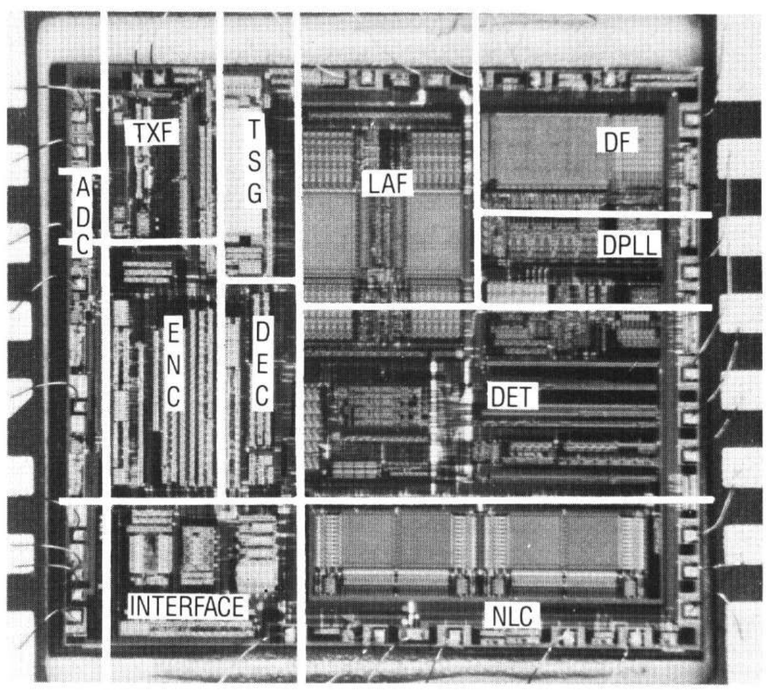
A single chip transceiver was designed collaboratively by BTRL and Mitel Semiconductors to meet the ANSI requirements for the U-interface in the integrated services digital network (ISDN). The device utilises echo cancellation and the 2B1Q line code to achieve high performance in the presence of near-end crosstalk and other impairments. Figure 7(*a*) shows a block diagram of the circuit. The digital signal processing (DSP) blocks of the circuit which perform the echo cancelling function were designed by BTRL and are contained within the dotted line on the diagram. All digital blocks except the DF operate at 5·12 MHz; the DF operates at 10·24 MHz. The remaining analogue blocks were designed by Mitel. Layout for the BTRL blocks, which contained about 50 000 transistors, was constructed to meet tight floor-planning constraints and transferred to Mitel for the final assembly of the total chip. Mitel was then responsible for manufacturing the chip on their 2 μm single-metal, double-polysilicon process and completed chips were tested by both organisations.

The collaborative nature of the project and the geographical separation of the design teams made special demands on certain aspects of the design methodology. Design for test, for example, took on special importance. It was essential that BT engineers could test their part of the circuit with minimum knowledge of the Mitel analogue blocks that lay between the digital core and the pins of the circuit. The DSP section was divided into the five blocks shown on the right of Figure 7(*b*) and a scan path was provided to enable control signals passing between the blocks at the symbol rate to be controlled and observed. Test control of high-speed bit-serial signals within the DSP is achieved by multiplexers at block inputs. All parts of the DSP functioned correctly first time.

Much of the early work on the 2B1Q design was done in conjunction with another silicon supplier and was intended for manufacture on this supplier's process. When Mitel became a BT subsidiary, and the decision was taken to develop a single chip 2B1Q interface, all the symbolic data generated using ASTRA could be readily translated into layout for the Mitel process.



(*a*) Circuit



JC: Jitter compensator
LEC: Linear echo canceller
NLC: Non-linear corrector
DF: Decimation filter
DPLL: Digital phase-locked loop
TSG: Training sequence generator
DEC: Decoder
DFE: Decision feedback equaliser
LAF: Linear adaptive filter
DET: Detect block
ADC: Analogue-to-digital convertor
TXF: Transmit filter
ENC: Encoder

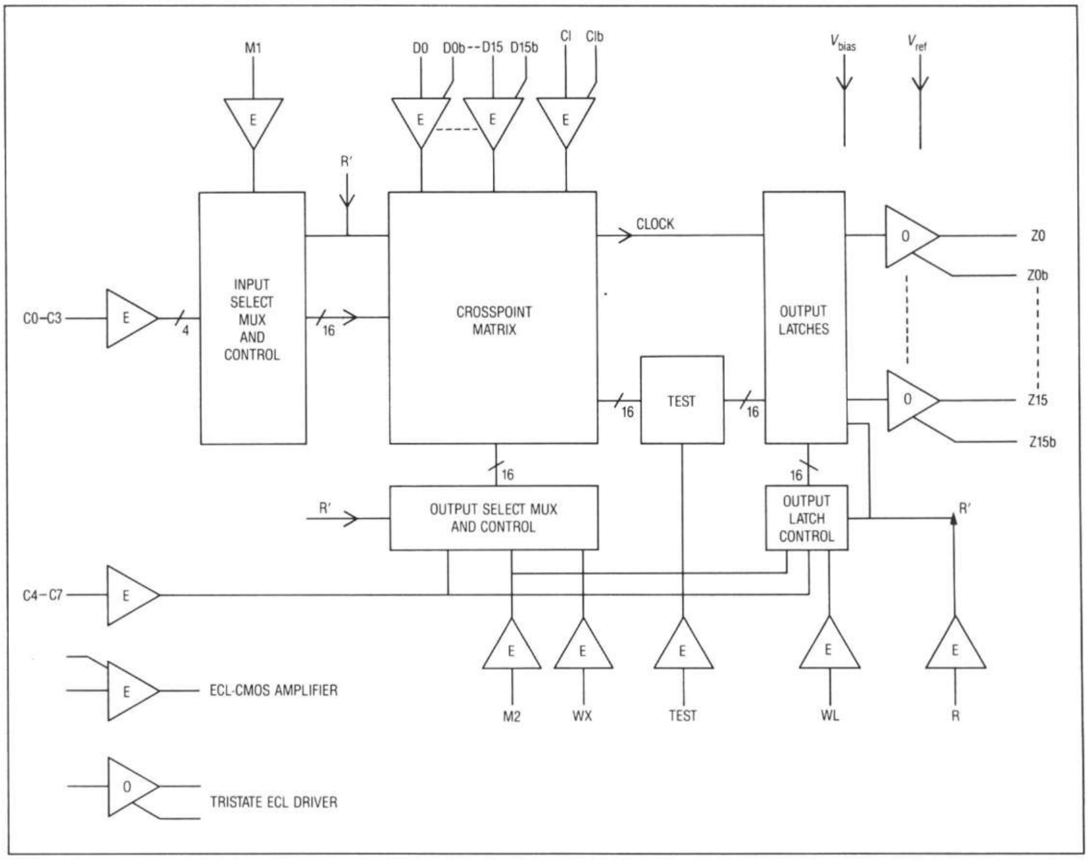
(*b*) Chip

Figure 7—2B1Q transceiver

CURRENT EXPERIMENTAL DESIGN WORK

Much of the expertise to develop the product chips described in the previous section grew from earlier experimental and prototyping studies, and the design work described in this section which is of an experimental nature is the essential

Figure 8
16 × 16 crosspoint



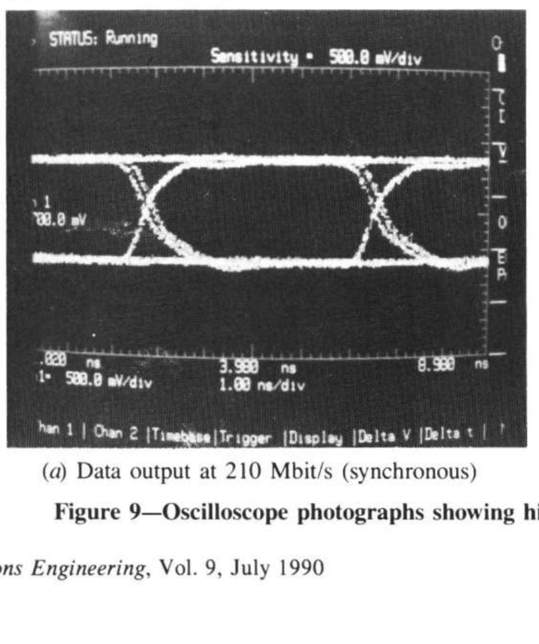
groundwork for future product and system developments.

16 × 16 Crosspoint—Very High Speed CMOS

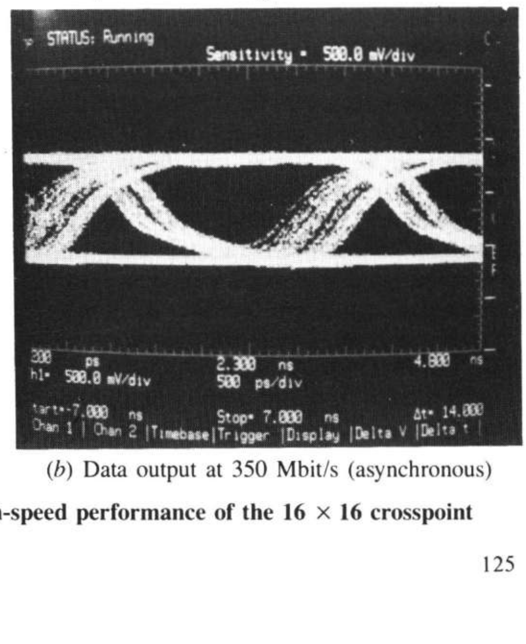
A 16 × 16 non-blocking digital switch was designed in the 0·7 μm double-metal CMOS technology developed by BTRL. This technology was optimised for high-speed performance, at the expense of high density, with a view to developing a prototyping capability for circuits operating up to 160 MHz and beyond. The drawn gate length of the transistors is 0·7 μm for n-channel and 1·0 μm for p-channel, but the metal pitch and other design rules are more typical of a 2·0 μm process.

The crosspoint, see Figure 8, was designed for a worst-case data rate of 155 Mbit/s in synchronous mode and the design estimate for power dissipation at 155 Mbit/s was approximately 1 W. The device has ECL-compatible differential inputs and outputs and operates from a single −5·2 V power supply. The chip is 6·2 × 5·9 mm in size and contains approximately 10 000 transistors. Outputs can be individually programmed, disabled or set into broadcast mode. Both synchronous and asynchronous modes of operation are supported and provision of tri-state outputs allows for cascading of the devices to create larger switch arrays.

The device worked first time meeting both logical and performance specifications. The oscilloscope traces in Figure 9 demonstrate syn-



(*a*) Data output at 210 Mbit/s (synchronous)



(*b*) Data output at 350 Mbit/s (asynchronous)

Figure 9—Oscilloscope photographs showing high-speed performance of the 16 × 16 crosspoint

chronous operation at 210 Mbit/s and asynchronous operation at 350 Mbit/s.

Wafer Scale Integration—Very High Complexity CMOS

BTRL is involved in an ESPRIT project to develop a wafer scale array of single instruction multiple data (SIMD) processors. Because all processed wafers have at least a few defects, wafer scale circuits use redundancy and reconfiguration to obtain an acceptable yield of working circuits. Provision of redundancy and reconfiguration is simpler on circuits, like the SIMD processor, that use arrays of identical circuit elements. Main applications for the SIMD device are in image processing.

In BTRL's implementation of wafer scale, the wafer is built up from multiple placings of a reticle, as for a wafer of conventional ICs, except that in this case the reticles abut to provide the interconnection between the adjacent sites. There is also an additional wafer-level metallisation for connecting from the edges of the peripheral reticles to bonding pads.

Each reticle contains four placings of a chip with 6 × 12 processing elements (PEs) plus reconfiguration switches and interconnection busses, a block of common-control circuitry and drivers for the bonding pads. Within each chip, there is another level of reconfiguration, consisting of a spare row and reconfiguration circuitry to by-pass the faulty PEs. The PEs are single-bit processors with 128 bits of RAM each and the instruction and RAM addresses are decoded on each chip from global command busses.

The basic PE consists of a full adder/subtractor, a number of latches and multiplexers, and two independently addressable 64 bit RAMs. Within a chip, each PE can communicate bi-directionally with each of its neighbours (north-south and east-west) and unidirectionally within a column through a communication register. The multiplexers allow different sources to be selected for the adder inputs and RAMs. A flag control enables individual PEs to be instructed to ignore the current instruction.

Reconfiguration takes place at two levels. Firstly, each chip operates a pseudo self-test from globally applied instructions and the result is held in a latch. The latch is interrogated using an electron beam tester and faulty PEs are bypassed. The reconfigured chip is retested and a 'good chip' map defined. The reticle level reconfiguration switch network is then tested to determine which switches function correctly and these results are combined with the good chip map to determine the best configuration for the final array.

The design is being implemented using a two-level metal, 1·2 μm CMOS process. The wafer level metal is part of the second-level metallisation, but is defined on a separate process mask. The array chip and reticle latch block were designed at BTRL using the ASTRA design system, and the final reticle, which was assembled at BTRL using the Cadence SDA design system, incorporated 'foreign' blocks from other partners.

MOSAIC III—Multi-Project Full Custom Bipolar Design

The high-speed MOSAIC III bipolar process from Motorola, described in the section on silicon process technologies, has been used by the high-data-rate IC design group for a number of different experimental designs. The designs were processed together on a multi-project wafer to reduce the engineering costs for prototype samples. The overall objective of the project was to assess the capabilities and limitations of the MOSAIC III process, the design methods and the interface to the foundry. Proven circuit configurations were used, as low risk options, alongside novel designs and new device geometries.

The circuits fabricated included analogue and digital functions, some examples being:

(*a*) OR/NOR, AND/NAND, XOR/XNOR;
(*b*) D-type flip-flop;
(*c*) 2:1 multiplexer, 16:1 programmable multiplexer;
(*d*) ECL/CML output buffers;
(*e*) automatic gain control amplifier, limiting amplifier, trans-impedance amplifier;
(*f*) phase detector, phase shifter, peak detector;
(*g*) multiplier;
(*h*) trans-impedance receiver;
(*i*) $2^{15} - 1$ pseudo-random binary sequence generator/detector; and
(*j*) $2^{7} - 1$ scrambler.

All designs tested are functionally correct and parametric measurements agree with simulations to ±10%. The D-type flip-flops have been demonstrated at data rates in excess of 3 Gbit/s and the output buffer stages, for interfacing to ECL 10K levels, operate at rates of 3·5 Gbit/s. Some of the circuits were used to construct a 2·4 Gbit/s data regenerator as shown in Figure 10. The regenerator can be seen as two distinct paths: the data and the clock extraction path. The circuits shown as shaded in Figure 10 were implemented using the MOSAIC III process. In the data path, the received signal at <10 mV is amplified by the automatic gain control (AGC) and the limiting amplifiers to a level suitable for re-timing. The clock extraction path consists of an analogue multiplier which generates a frequency component at 2·4 GHz from the non-return-to-zero (NRZ) data. The clock frequency component is filtered using a combination of strip-line and surface acoustic wave (SAW) filters. A limiting amplifier compensates for the insertion loss of the filters and provides a clock signal at 2·4 GHz for re-timing of the data. The limiting amplifier is designed

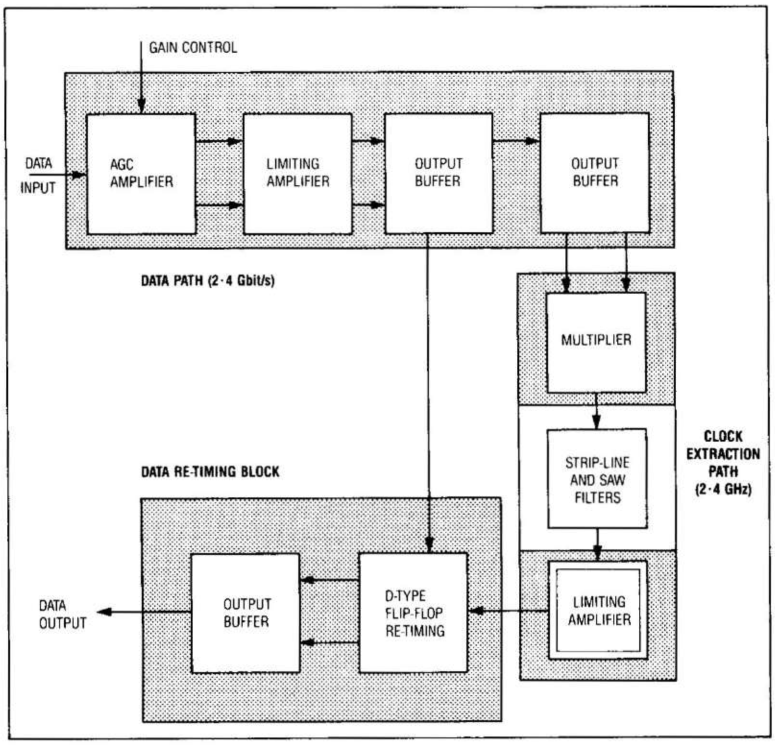


Figure 10
2·4 Gbit/s regenerator block diagram

for low amplitude to phase conversion. Finally, the amplified data is re-timed using the extracted clock signal.

This technology will be of practical use in high-bit-rate optical-fibre transmission systems for international and domestic routes as well as local loop developments. Other application areas are high-speed local area networks and trunk crosspoint switching.

CONCLUSIONS

This article has described some of the recent silicon design work carried out at BTRL and given a glimpse of the experimental designs that will enable more complex and higher-performance systems and products to be developed in the future.

Paradoxically, the ubiquity of silicon IC technology sometimes obscures its continuing importance in driving forward today's developments in telecommunications and information technology, but it will continue to have a high strategic value for major companies in these businesses.

Sub-micron silicon processes are making the manufacture of systems with over a million transistors on a chip possible and wafer scale integration could take us into the realm of systems with tens of millions of transistors on a single piece of silicon.

Manufacturing technology will not be the limiting factor; managing the design complexity will determine what is possible. New CAD tools will be essential, and the new design methodology trends are being established now. Human intelligence will be used for the higher levels of the design process (behavioural and functional) and automated tools will be used increasingly for the lower levels (logic and layout). But for most of the next decade there will be a need for silicon designers who understand and appreciate the design problems at the physical level to obtain competitive levels of integration and optimum levels of performance from silicon integrated circuits.

ACKNOWLEDGEMENTS

The author would like to acknowledge the assistance of many colleagues in the IC design section of BTRL in the preparation of this article. Further assistance came from the Product Engineering and Technology Division in the fabrication and testing of the crosspoint circuit, from the Optical Networks Division on some of the MOSAIC III circuits, from the Switched Networks Division on the 2B1Q transceiver and from the Corporate Programme Office for the advanced development work.

Biography

Alex Elliot joined the Post Office Research Laboratories at Dollis Hill after obtaining a B.Sc. degree in Physics at Edinburgh University in 1960. He started his career working on thermionic emission and electron physics investigations for vacuum tubes used in submarine transmission systems. This was followed by a period in which he was engaged in a series of studies concerned with the electronic properties of semiconductor materials, during which he obtained a Ph.D. from Imperial College for work carried out on the electrical properties of thin silicon films on sapphire substrates. Subsequent work interests have included process technology, reliability physics and test technology for integrated circuits and he is currently in charge of the Integrated Circuit Design Section at BTRL.

Technical Publications

M. LYNAS†

Technical publications play a key role in supporting British Telecom products. This article explains why these publications are so important to BT and gives an overview of the stages involved in the production of good publications. The article also briefly looks at electronic publishing and the alternatives to paper itself as a medium for technical publications.

INTRODUCTION

On 23 April 1983, the Technical Publications Unit (TPU) was formed, bringing BT in line with the majority of large companies world-wide for the first time.

Technical publications is the name widely used in industry to describe the instructional material that is supplied with a product or service. Some publications are produced for the benefit of the person using the product and these are usually called *user guides*. Other publications are aimed at the people responsible for installing or maintaining the product and these are known as *engineering manuals*. Technical publications should not be confused with sales brochures or internal company publications.

The process of explaining how to install, maintain or use a product is called *technical communication*. Good technical communication is vitally important in order to meet customers' requirements fully and minimise the cost of supporting the product. Whether achieved through training or through instruction books, the objective of technical communication is to help end-users understand the product or service they are using.

There are a number of professional bodies aiming to maintain standards in this field. In the UK there is the Institute of Scientific and Technical Communicators. Serving North America and the rest of the world there is the Society of Technical Communicators. The IEEE also has a specialist group devoted to professional communications.

TECHNICAL PUBLICATIONS IN BT

For many years, instructional material in BT was prepared in the form of Telecommunications Instructions (TIs) or Inland Services Information System (ISIS) documents. During the last 10 years, however, the number of products and services offered by BT has increased considerably and in particular the need for high quality user guides has grown. Managers throughout the company have consequently been faced with the task of arranging the preparation, printing and distribution of publications for their own products.

It was recognised that this task required a contribution from professional technical communicators and so the TPU was formed.

THE TECHNICAL PUBLICATIONS UNIT

The first task of the embryo TPU was to research BT's needs and apply the principles already established in the technical publications industry.

The core skill required to produce good technical publications is that of the technical author, and so a number of authors were engaged to suit the different types of products and publications required.

Other specialists in the fields of illustration, editing and electronic publishing were also recruited, and procedures were drawn up to show how technical publications should be prepared to fit in with the launch plan of the product.

From the outset, it was agreed that the greatest value that TPU could add was to ensure that BT's technical publications are always:

(*a*) available,
(*b*) tested as fit for purpose and easy to use,
(*c*) conformant to BT house style,
(*d*) distinctive,
(*e*) liked by customers,
(*f*) supported throughout the lifetime of the product, and
(*g*) good value for money.

During 1989/90, over 500 publications were processed by trained people in the TPU. Many of these people have had considerable experience of the technical publications industry in the UK and Europe.

IMPORTANCE OF GOOD TECHNICAL PUBLICATIONS

There are a number of popular misconceptions surrounding the preparation and use of instructional material. For example:

† Finance and Administration, British Telecom Research and Technology

(*a*) publications are not important,
(*b*) they are a necessary evil,
(*c*) they increase the cost of the product,
(*d*) anyone can write them,
(*e*) people only use them as a last resort, and
(*f*) BT should supply the minimum that it can get away with.

However, it is a known fact that technical publications are important to the success of a product and they can actually reduce the lifetime support costs.

The reason why instruction books are sometimes used as a last resort is partly because they have developed a poor reputation, partly because many people prefer to experiment and partly because many publications are in fact poorly written or overwhelmingly large.

It has been discovered, through FiX (Fault Information Exchange) reports and hotline calls, that bad publications can, and do, cost BT a lot of money. These costs are often overlooked as they are incurred in many different areas of the company. Often products are reported faulty when this is not the case, resulting in expensive call-outs for BT. It has also been found that bad publications annoy customers intensely and cause concern to District staff when, for example, poor maintenance instructions lead to longer out-of-service periods.

Product Liability

The cost of *not getting it right first time* can be multiplied many times when customers are injured, their property is damaged or business is lost as a consequence of inadequate instructions. The law on liability[1] places greater demands on suppliers than is realised and there are many legal precedents already established.

Characteristics of Good Publications

In order to avoid the problems that result from bad publications, it is first necessary to understand what makes a good publication. A good technical publication should have:

(*a*) a defined maximum number of pages,
(*b*) a good index and other 'navigational' aids,
(*c*) logical structure,
(*d*) correct balance of tutorial and reference information,
(*e*) simple words and short sentences,
(*f*) written style appropriate to the target users,
(*g*) no unexplained jargon,
(*h*) easy-to-read typeface and page layout,
(*i*) clear well-designed illustrations that complement the text,
(*j*) attractive appearance, and
(*k*) convenient format and size.

Most important of all, a good publication will allow users to complete key tasks within a defined period of time and meet their primary requirements.

Total Quality Management (TQM)

To achieve a dramatic improvement in the quality of BT's technical publications and reduce the cost of poor quality, the principles of total quality management (TQM) are being applied. The current problems are being analysed and the possible solutions and potential barriers are being identified. Probably the greatest barriers to achieving the necessary improvements are concerned with the timescales and budgets allowed for the preparation of publications. There is also the need to explain why trained professional technical communicators are required to prepare publications.

TQM is very much a *prevention culture* based on the principle that the earlier an error is detected, the smaller is the overall cost to the company. Redmill in 1988[2] has outlined a documentation inspection method to reduce the number of errors in publications.

As part of the education and communication stage, steps are being taken to ensure that a greater understanding of the importance of technical publications is achieved within the company, especially with respect to life-cycle costing. This requires closer links with the product management teams in the company.

Feedback from customers is already being monitored to assess the improvements that are being made.

PRODUCING GOOD PUBLICATIONS

Before undertaking any technical publications project, the necessary resources need to be organised. Qualified authors, editors, illustrators and keyboarders need to be available under the leadership of a project manager who is experienced in the fields of technical communication, typesetting and printing. Workstations equipped with word processing and electronic publishing software are required for the authors and keyboarders, in addition to a well-equipped studio for the illustrators.

A quality management system detailing the key procedures is an essential requirement for the team and will need maintaining on a regular basis. Training sessions for all members of the team to reinforce the procedures must also be arranged on a regular basis.

For the preparation of each publication, a project manager (PM) is assigned to take responsibility for all aspects from beginning to end. The PM creates a job file that will contain all the relevant correspondence and planning and control documents. In situations where there are large numbers of projects being controlled, it is an advantage to use a database to record key data such as estimated and actual hours, required-by dates and publication numbers. The computer can then search for publications meet-

ing certain criteria or carry out useful calculations such as cost to date.

Figure 1 shows the typical stages involved in the preparation of technical publications.

Researching and Analysing the Product

At the earliest stage in the development of a new BT product or service, consideration should be given to the technical publications requirements. Specialist resources may be required and there could be a significant learning curve for those involved.

A checklist is the most reliable way of ensuring that all the right questions are asked at the beginning of the project.

The questions that need to be asked most often are:

(*a*) how will the product be used and maintained,
(*b*) what are the target times for completing key tasks on the product,
(*c*) what is the expected lifetime of the product,
(*d*) how often will the product be updated,
(*e*) when will the design be frozen,
(*f*) when will a prototype be available,
(*g*) could the instructions be built into the product itself,
(*h*) can the product be designed so that use is more intuitive and fewer instructions are needed,
(*i*) who will use the product and what will their background and experience be,
(*j*) what training will be given to customers and BT staff,
(*k*) what are the needs of the Districts,
(*l*) what kind of publications are needed,
(*m*) what volume of publications is likely to be required,
(*n*) what is the target number of pages for each publication,
(*o*) are foreign language versions required,
(*p*) what is the target price of the publication,
(*q*) what are the key milestones of the project,
(*r*) what are the arrangements for testing the publications with the product,
(*s*) what source material is likely to be available,
(*t*) how will feedback from end users be obtained and used once the product is in the field, and
(*u*) will the publication be frequently updated?

The quality of the final publication depends greatly on the quality of the answers to these questions.

Planning the Project

The first stage in planning a technical publication project is to obtain a specification of requirements. The project manager then considers how the publication or family of publications is to be produced. There are several ways in which this can be done. The manufacturer of the product may have facilities for preparing technical publications and may be able to produce publications to BT style specifications for the lifetime of the product. Another way is to engage the services of a free-lance author or a technical writing agency ensuring first that they are up to the standards required by BT. TPU helps product managers to decide on the best alternative.

Once the method has been decided upon, resources are scheduled and the cost of the project is estimated. This is made up from the cost of:

(*a*) labour, materials,
(*b*) subcontracted work such as typesetting and printing,
(*c*) storage and distribution costs, and
(*d*) contribution to overheads and accommodation.

After agreeing the budget for the project with the product manager, work can begin.

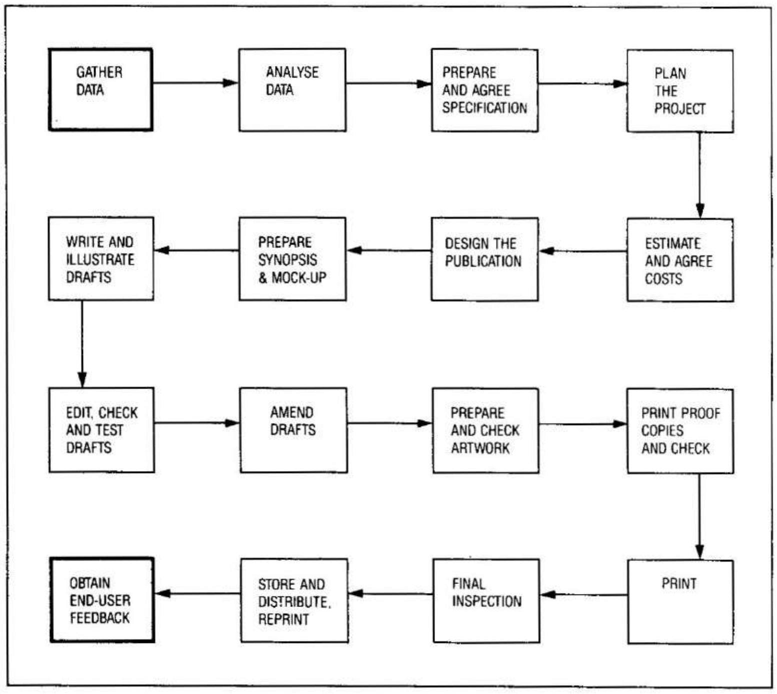


Figure 1
Typical steps in the preparation of technical publications

Designing the Publication

Sometimes it is better to split a publication into different parts in order to avoid producing a book which is large and awkward to use. A family of publications can therefore be produced such as a tutorial booklet, a reference manual and a quick reference card.

The size, format, method of binding and packaging can all affect the cost and usability of the publication and should be considered carefully. Illustrators should be consulted by project managers at the design stage to advise on the appearance of the publication including the cover and the use of illustrations, photographs and graphical devices on the pages inside.

The choice of typefaces, use of colour and the page layouts are dictated largely by the requirements of the BT style guidelines. A mock-up of the finished publication can be prepared if necessary to give an idea of the final appearance and effect.

Writing and Illustration

A synopsis is often prepared to give the product manager an indication of the content of the publication especially if the publication is large or complex. A synopsis is essentially an expanded contents list.

TABLE 1
British Standards and other Reference Documents

The Consumer Protection Act, 1987. (Published by HMSO)	
The Classification, Packaging and Labelling of Dangerous Substances Regulations, 1984. (Published by HMSO)	
Specification for page sizes for books	BS 1413: 1970 (1982)
Glossary of paper, board, pulp and allied terms	BS 3203: 1979
Recommendations. The preparation of indexes to books, periodicals and other publications.	BS 3700: 1976 (1983)
Specification for sizes of paper and board.	BS 4000: 1983
Glossary of terms used in offset lithographic printing.	BS 4277: 1968 (1986)
Specification for technical manuals.	BS 4884
Part 1: 1973 (1983) Content	
Part 2: 1974 (1983) Presentation	
Copy preparation and proof correction.	BS 5261
Part 1: 1975 (1983) Recommendations for preparation of typescript copy for printing.	
Part 2: 1976 Specification for typograhic requirements, marks for copy preparation and proof correction, proofing procedure	
Safety signs and colours	BS 5378
Part 1: 1980 Specification for colour and design	
Part 2: 1980 Specification for colorimetric and photometric properties of materials	
Part 3: 1982 Specification for additional signs to those given in BS 5378: Part 1	
Glossary of documentation terms	BS 5408: 1976
Recommendations for loose-leaf publications	BS 5641: 1982
Specification for quantities, units and symbols	BS 5775
Parts 0–13 1979–1982	
Recommendations for the presentation of tables, graphs and charts	DD 52: 1977

For further information contact: BSI Enquiry Service, Linford Wood, Milton Keynes MK14 6LE. Tel: 0908 221166.

British Standard BS 4884 gives an outline of what to include in a publication to cover all eventualities. Table 1 lists other relevant British Standards. It is at this stage that the structure of the publication is considered to ensure that the end result is logical and complete.

The next stage is the writing of the publication, and this is usually directly on a word processor. At all times during the writing of the publication, the reader must be borne in mind by the author. In BT, the recommended software for this purpose is WordStar™ and Microsoft Word™. The correct written style for user guides employs the active voice and the second person. This style is less intimidating and is easier to read than other styles; for example, 'When you have keyed in the telephone number ...' is written instead of 'After keying in ...'.

Simple words should be used wherever alternatives exist, and long sentences should be avoided because they are difficult to read. Much research has been carried out to assess the factors that affect the readability of publications. The best known readability measures of a piece of writing are Gunnings fog index and the Flesch index. The fog index of a technical publication, like that of a tabloid newspaper, should be low to make it easy to understand for the majority of readers.

The Flesch index is defined as:

$$206{\cdot}835 - \left(\frac{\text{number of syllables}}{\text{number of words}} \times 84{\cdot}6\right) + \left(\frac{\text{number of words}}{\text{number of sentences}} \times 1{\cdot}015\right).$$

Gunnings fog index is defined as:

$$(\text{average sentence length} + \text{percentage of long words} \times 0{\cdot}4) + 5,$$

where long words are taken to be those with more than two syllables. The result is the equivalent UK reading age.

All technical terms that are used must be explained in a glossary and anything that the reader may need to refer to should be included in the index or table of contents. In general, all publications of more than 20 pages should have an index. There are various software packages that allow indexes and tables of contents to be created automatically.

The first draft is the starting point for checking the effectiveness of the publication and for identifying further refinements. Usually, there is a second draft and a third or final draft. However, for extended or complex projects, there could be many iterations before the final draft stage.

Illustrating a technical publication is an integral part of the design process. Line drawings, diagrams and photographs may all be used to complement the text. It is important that the illustrations are clear and positioned correctly in the final publications. Illustrations that are not

on the same page as the text that refers to them are potentially misleading and mean extra work for the reader. Like technical writing, studio work is labour intensive and sufficient time and budget must be allowed. The investments made, however, will be more than paid back later in reduced support costs for the product.

Editing, Checking and Testing

At each draft stage, the publication is edited and checked for style, grammar, spelling and accuracy. The proof correction marks shown in Figure 2 are used. Checking spelling is now a very simple task if the appropriate software is used. The readability of a publication can also be checked easily using inexpensive software on a PC.

The usability of a publication must be checked at an early stage to ensure that users do not have any difficulties in practice. The most effective way of doing this is by video recording a typical user trying to use the product and the publication together.

The user is encouraged to think aloud so that reactions may be recorded. This technique has been used by TPU successfully on several publications using a basic test room. The process also highlights potential problems with the product and allows design improvements to be considered.

The other method of testing the usability of a publication is to use the alpha and beta trials of the product as an opportunity to obtain feedback from users. Interviews with users and carefully designed questionnaires are used to measure the success of the publication.

marginal mark	textual mark	meaning
/	none	end of this amendment
⅄ followed by new text	⅄	insert text indicated in margin
	/ through character(s) or ⊢⊣ through words	delete the characters(s) or word(s) marked
new word(s) character(s)	/ through character(s) or through words	substitute character(s) or word(s)
⊗	circle around characters	wrong font
▬	▬ under character(s)	change to upper case
～～	～～ under character(s)	change to bold type
	circle around characters	change to lower case
⊙	/ through character or ⅄ where necessary	insert or substitute full stop
	/ through character or ⅄ where necessary	insert or substitute hyphen
		new paragraph
		run on
	between words or characters	transpose words or characters
		indent
		cancel indent
		take over
	between words or characters	close up words or characters
Y	Y between words	insert space between words
or		insert space between lines or paragraphs
or		reduce space between lines or paragraphs

Figure 2
Proof correction marks as recommended in BS 5261: Part2: 1976
(courtesy British Standards Institute)

Artwork Preparation

Once a final draft has been produced that meets the approval of the product manager and which has been tested satisfactorily, work can begin on producing *camera-ready artwork* (CRA). This is so called because the final made-up pages are photographed by the printer for offset litho printing or used as masters for photocopying. There are two main ways of producing CRA representing two distinct levels of appearance quality.

The first is to use one of the various desktop electronic publishing (DTP) software packages that run on PCs. The system used by TPU and many other parts of the industry is called *Ventura Publisher®* .

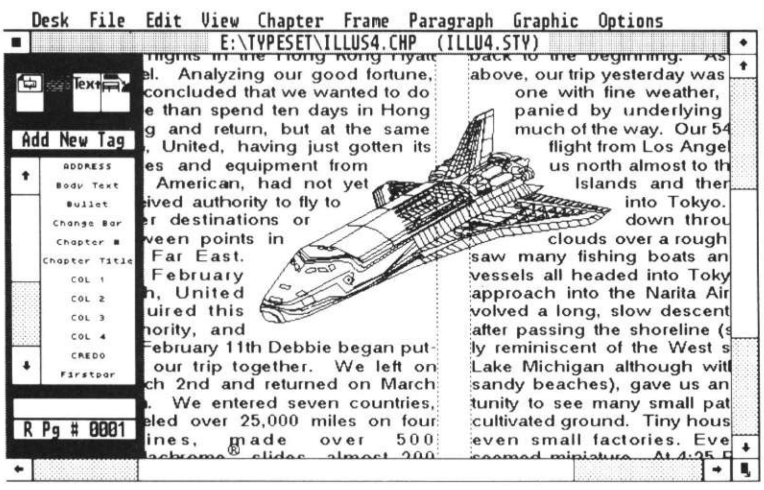
Typically, the word processor file representing the final draft is read into the DTP system and typesetting commands are then used to mark the text in order to achieve the correct page layout and typographic style. Most DTP systems have a 'what you see is what you get' (WYSIWYG) facility which means that what is seen on the screen of the PC is what the printed page will eventually look like. Figure 3 shows a typical screen. The output of a DTP system is via a laser printer or via conventional typesetting equipment.

Desktop laser printers are a convenient and relatively inexpensive way of producing CRA, but the quality is currently not as good as typesetting because of the limited resolution. Most laser printers available today are capable of resolving no more than about 300 dots per inch whereas typesetters are capable of at least 1000 dots per inch. This situation is changing rapidly, however, and 400 × 800 dot lasers are now becoming commonplace.

The second option is to send the word processor file to a typesetting agency either by floppy disk in the post or via a modem link. Output from the typesetter is then posted or faxed to the project manager for checking.

Illustrations may be scanned and stored electronically or they may be pasted in conventionally. The pasting up of artwork is becoming less common as technology allows faster scanning, easier manipulation, faster printing and greater storage capacity on computers.

Figure 3
Ventura desktop publishing software



Checking the final artwork is very important because this is the last chance to avoid errors before the costly business of printing begins. Using an automated DTP process will reduce the chances of errors creeping in and eliminate the possibility of pasted-up work falling off. It is still quite possible, however, to inadvertently delete text or simply to forget to include sections of text.

When fully checked, a copy of the camera-ready artwork is sent to the product manager for final approval.

Printing and Finishing

The camera-ready artwork is photographed and separate films are produced for each colour required in the publication. The films are then used to produce *plates*. Plates, usually the metal kind, are used for long print runs and are made using a photographic and chemical etching process. Some printing and typesetting companies have facilities for etching plates directly using laser technology.

The plates are fastened to the drums on the printing press and inked up in order to transfer the image of the page onto paper. A different plate for each colour is required for each page or group of pages. The printed pages are checked for quality on an almost continuous basis as they emerge from the press and are then collated to form the finished publication. The finishing process includes folding, guillotining the edges of the pages, binding the pages, and fitting the covers.

The choice of binding method is dictated by a number of different requirements. Wire binding allows books to stay open easily and means they can be folded back to take up less desk space. Another method uses ring binders which are useful for publications that need frequent updating.

The printed publications are delivered to the product manufacturer for packing with the product or sent to one of the BT warehouses for storage and subsequent call off.

Publication Control

Printing the technical publications is not the end of the story. The publications must be controlled during the entire life cycle of the product, especially if they are part of an approved quality system. Each publication requires a publication number and an issue number to enable the publication to be associated with a particular release of the product. If the publication is a controlled document, holders are registered with the appropriate project manager who will ensure that updates and amendments are sent out promptly.

A helpdesk for the use of Districts and customers allows user feedback to be collected, queries to be answered and requests for additional copies to be processed efficiently. This is also the point to which end-user questionnaires are sent for analysis.

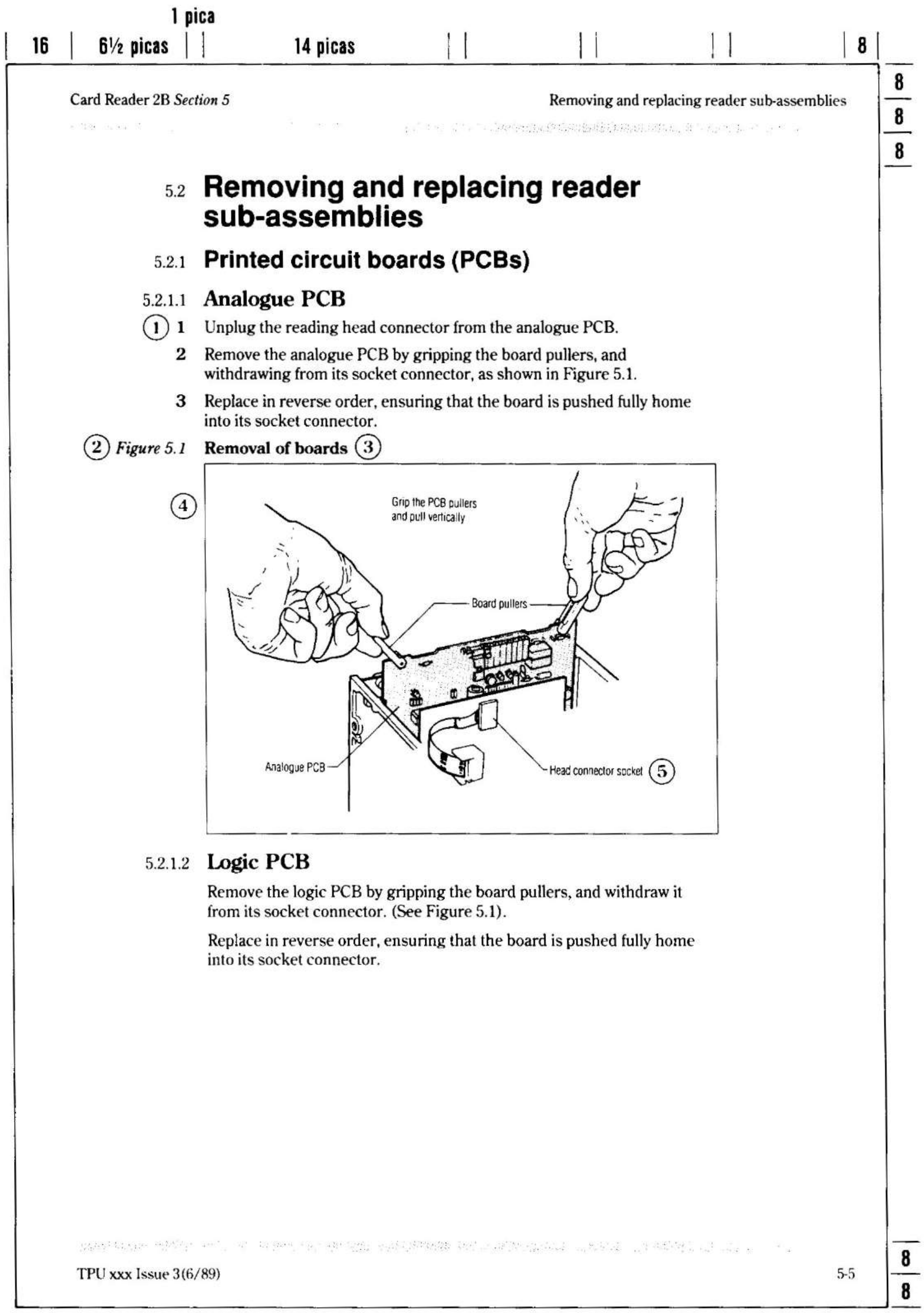
STYLE AND PREPARATION GUIDELINES

The Product Style Initiative (PSI) team in BT is preparing a number of guidelines that describe the style of BT products, packaging and documentation. In future, all technical publications will have to conform to these guidelines to ensure that BT instructional material has a consistent image and meets defined standards of quality.

TPU has been involved in the preparation of these guidelines which cover the following aspects of technical publications:

(*a*) planning projects,
(*b*) designing and producing publications,
(*c*) use and preparation of illustrations,
(*d*) editing and testing,
(*e*) typographic design,
(*f*) print specification, and
(*g*) documentation control.

An example of the PSI approved page layout is shown in Figure 4. Typefaces and sizes have been chosen on the basis of readability and what has already been selected for promotional ma-



Card Reader 2B *Section 5* | Removing and replacing reader sub-assemblies

5.2 **Removing and replacing reader sub-assemblies**

5.2.1 **Printed circuit boards (PCBs)**

5.2.1.1 **Analogue PCB**

(1) **1** Unplug the reading head connector from the analogue PCB.

2 Remove the analogue PCB by gripping the board pullers, and withdrawing from its socket connector, as shown in Figure 5.1.

3 Replace in reverse order, ensuring that the board is pushed fully home into its socket connector.

(2) *Figure 5.1* **Removal of boards** (3)

(4)

5.2.1.2 **Logic PCB**

Remove the logic PCB by gripping the board pullers, and withdraw it from its socket connector. (See Figure 5.1).

Replace in reverse order, ensuring that the board is pushed fully home into its socket connector.

TPU xxx Issue 3(6/89) | 5-5

Figure 4
Typical page in A4 house style approved by BT

Key:

1. Numerical listing
11 pt Century Old Style Bold figures

2. Figures and table references, notes and examples
11 pt Century Old Style Italic upper and lower case

3. Figure and table titles
11/13 pt Century Old Style Bold upper and lower case

4. Illustrations
Contained within a 0·5 pt ruled boxed area

5. Legends
9 pt Helvetica 57 upper and lower case

terial in BT. The page layout is based on results of the work carried out by Professor James Hartley of Keele University[3] into the design of instructional text. Line lengths, inter-line spacing, heading design and use of white space are all factors that affect readability and are therefore specified.

The BTUK product management process also provides for technical publications to be produced as part of the launch plan.

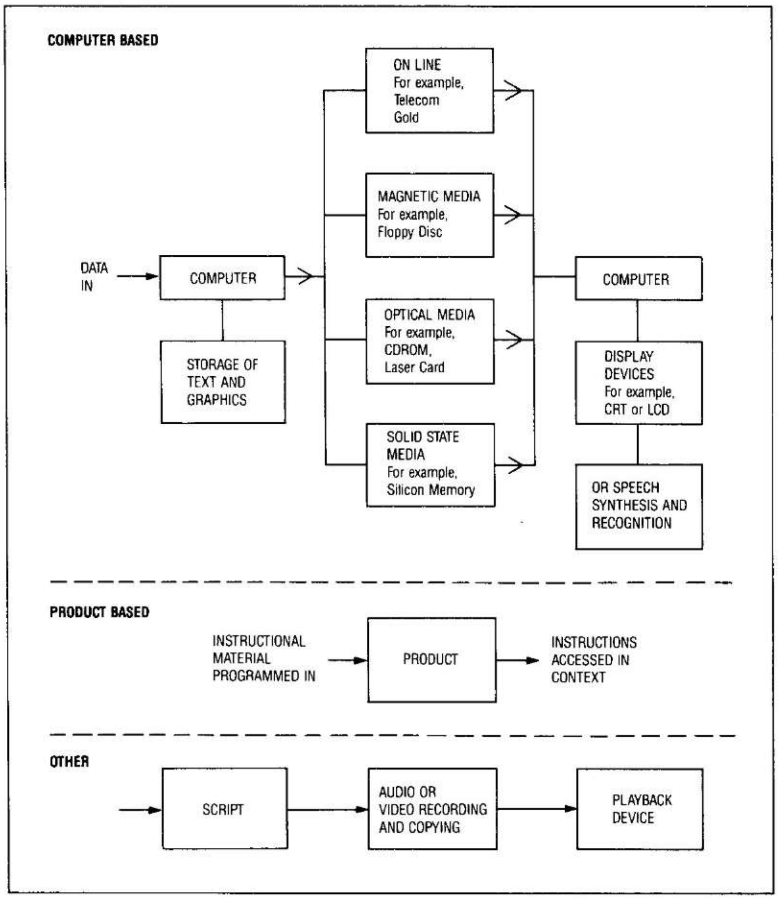
ELECTRONIC PUBLISHING

The term *electronic publishing* is used in different industries to mean different things. Usually it is used to describe the process whereby paper is printed electronically, usually via laser printers. Desktop publishing is a compact and relatively inexpensive form of this process and has been described earlier as part of the page make-up procedure.

Larger, corporate electronic publishing systems using mini or mainframe computers and high-volume laser printers have been in common use in large companies for several years. These systems cost around £500 000, but have the advantage of allowing printing to take place on demand, avoiding the need for warehouses and eliminating the problem of obsolete stock.

There is a trade off between electronic publishing and offset litho printing with respect to cost. Offset litho is much cheaper for long print runs of about 1000 or more.

Figure 5 Alternatives to paper as a medium for technical publications



Sometimes, electronic publishing is used to describe the process of supplying information without actually printing on paper at all. On-line databases such as Jordans and Infocheck are examples of this kind of publishing.

Of course, electronic publishing can be used for all kinds of information, not just technical publishing.

THE FUTURE

In the context of technical publications, there are several alternatives to paper as a medium for the communication of information and many of these have been used in recent years. Figure 5 shows these different ways schematically. The earliest alternative to paper was the compact audio tape cassette which was, and still is, used to give a tutorial or introduction to the product. This method is not suitable for providing reference information because of the difficulty of accessing particular data. The video cassette has been used similarly and, while having the advantages of a visual medium, is more expensive to produce and requires playback facilities which may not be readily available.

The 'electronic' alternatives to paper offer many advantages over the paper or audio/visual media and can potentially use the same word processor files as a source of data. Different products and services require different approaches. For example, for a screen-based PC, the simplest alternative to paper is the provision of a floppy disk containing the exact contents of the user guide. The user can then use a search facility to rapidly find the information that is required. It is generally accepted that the readability of screen-based text is significantly worse than paper-based text making this method less suitable for tutorial information. To improve the effectiveness of this method, the information in the user guide should not be transferred verbatim onto the computer, but structured and presented in a more dynamic way. Flashing warnings and contextual help screens are examples of this better approach and indicate a growing integration of product and instructional material.

Another example of how an alternative to paper could be more effective is the use of liquid crystal displays on telephones. Details of the facilities available on a direct line or extension telephone and how to use them can be stored within the instrument itself or in the central processor. There are several telephone systems available today that do this and give helpful prompts when various facilities are used.

It can be seen that products are becoming increasingly intelligent and a wide range of possibilities is emerging which includes computer-based training, voice synthesis, and expert documentation systems which provide electronic navigation.

CONCLUSIONS

Technical publications are essential for the successful launch and support of every BT product and service. They require a significant investment in time and money as well as professional planning, production and evaluation. Advances in technology are making it increasingly possible to integrate instructions with products leading to significant benefits for BT customers.

ACKNOWLEDGEMENTS

The author wishes to express his thanks to those who have assisted in the preparation of this article and to all the people in TPU who have enabled the Unit to become established in BT as a centre of expertise in the field of technical publications.

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Biography

Martin Lynas joined the company in 1968 as an open competition Assistant Executive Engineer and worked briefly in the North East Region and then with Satellite Systems Development before joining the BT Research Laboratories. In 1983, he moved to what is now Communications Systems Division to set up the Technical Publications Unit.

AXE10: Ready to Connect

A. G. S. PAPASPYRU, G. STANLEY, and S. HAWKINS†

The development of AXE10 to meet BT installation requirements has led to the evolution in system testing and delivery quality that has applications beyond the original objective. This article describes the AXE10 factory pre-tested equipment and its use in BT installation applications, such as turn round of small exchanges with little spare space and outdoor cabinet exchanges.

INTRODUCTION

In March 1985, British Telecom placed a contract with Ericsson for the supply of the AXE10 digital exchange system. AXE10 was chosen as the second digital system in the BT network to introduce competition into the digital local exchange modernisation programme, which up to that time was exclusively provided by GEC and Plessey System X exchanges.

The initial contract placed covered a two year ordering period. The first year comprised some 100 000 lines and the second year had a planned minimum order of 300 000 lines. Since placing the original contract, there have been extensions to the contract period, and, half way through the fifth year programme, total orders are approaching 3 million lines.

All the initial orders for AXE10 were for supply and installation by the contractor. However, at the time of placing the contract, BT was heavily engaged in a large programme of System X direct labour installations by BT staff. Once the initial AXE10 orders were placed, attention was focused on providing the capability for BT installation of AXE10. This article records the development of the factory pre-test AXE10 equipment to meet this need, and describes deployment options in the BT network.

AXE10 INSTALLATION

As digital switching systems, AXE10 and System X have many architectural similarities. However, significant differences exist in the method of commissioning exchanges. This difference was the genesis of the factory pre-test equipment concept and the key to other off-shoot opportunities.

This fundamental difference in installation technique lies in the commissioning of remote concentrators. Commissioning of System X remote concentrator units (RCUs) can be carried out without connection to the host processor unit, and, by use of special test equipment, the exchange can even be demonstrated to, and accepted by, BT without connection to the host. Installation of the AXE10 remote subscriber switch (RSS) does not make use of such test equipment, and the commissioning and demonstration of the exchange require connection to the host processor site. Therefore, if BT is to carry out installation, the requirement for AXE10 RSSs to be connected to the host processor has a significant impact.

To commission/install an exchange requires that the installation staff are trained to the necessary level. For the standard installation technique of an AXE10 RSS, a minimum of one year of training and field experience is required to provide sufficient competence to do the work without close supervision. The need for BT to invest one year of training per man was not compatible with the probable level of work arising from BT installation opportunities. A small RSS installation typically takes only 6 weeks. It was clear that unless the amount of training required could be significantly reduced, there was no economic possibility of meeting the demand for BT direct labour installation.

The requirement for this high level of training was analysed by the AXE10 supplier, Ericsson. It was found that a significant reduction in on-site testing could be achieved by testing complete exchanges in the factory and then shipping the equipment to site. On site, a simple installation procedure and verification test would then be sufficient to bring such equipment into service. This approach was trialled at four field sites in 1987 and its success has led to a significant number of supply-only exchanges being ordered. The factory pre-test process and the necessary modification to the equipment are described in the remainder of this article.

The concept of testing exchanges in the factory is not restricted to conventional exchange systems. AXE10 can also be delivered in weather-proof outdoor cabinets, and the first of these was delivered to BT and brought into service in the summer of 1988.

FACTORY PRE-TEST

The AXE10 RSS consists of hardware and software modules that provide line capacity for up to 2048 lines. A number of RSSs can be combined to form a remote concentrator centre.

† Network Systems Engineering and Technology, British Telecom UK

Within the RSS, the total hardware package is known as an *extension module group* (EMG). The EMG is the basic unit on which factory pre-test is based.

In the factory pre-test process, the EMG is assembled from cabinets, standard cable sets and magazines in accordance with each specific District order. Figure 1 shows a factory-cabled cabinet ready for magazine loading. The completed EMG is then passed into the test process, which consists of both functional and environmental tests.

Testing takes place at the test plant in the Ericsson factory at Scunthorpe. The complete EMG is placed in a heat chamber where the temperature is raised to 50°C and basic functional testing at the 2 Mbit/s level, is carried out over an 18 hour period. The temperature is then reduced to 45°C and full functional testing is performed, concluding with a load test using a high-capacity traffic tester generating up to 200 calls per minute for 12 hours. Faults found are cleared and testing continues until the call failure rate can be held below the maximum level of eight in 10 000.

A comprehensive quality check is made of the entire process from construction of the cabinet to testing the EMG, and a quality assurance certificate is issued for each cabinet.

It is important to note that once functional testing of each EMG has been completed, disturbance of electrical connections, from factory to site, is kept to a minimum. Only external test connections and inter-cabinet bus cables are removed prior to separating the cabinets for delivery to site. The bus cables, having already been tested with the EMG as part of the pre-test, are labelled and shipped to site to be re-connected in their prescribed tested positions.

One of the criteria for factory pre-test is the minimum disturbance of cables once testing is completed. Conventional installation of AXE10 requires termination of the main distribution frame (MDF) cables directly onto the line cards. To minimise the number of cables and plugs to be disconnected after the pre-test, the connections to the line cards are brought out to a plug field to which the test lines and the MDF cables are connected. The plug field can be seen on the left hand side of the cabinet in Figure 1.

The magazines of PCBs are firmly located in very sturdy cabinets, and lorry transport employs air suspension; therefore, very little packaging is required for transportation of equipment from factory to site. Cabinet edges are protected with stout cardboard and a large polythene bag is placed over the cabinet to keep out dust and damp. See Figure 2. A modified pallet lifter, incorporating side supports and a brake, is used to wheel cabinets from factory floor to lorry then from lorry into buildings through normal 'pedestrian' doorways. See Figure 3. Access to upper floors can be gained by using passenger lifts or hoist as available.

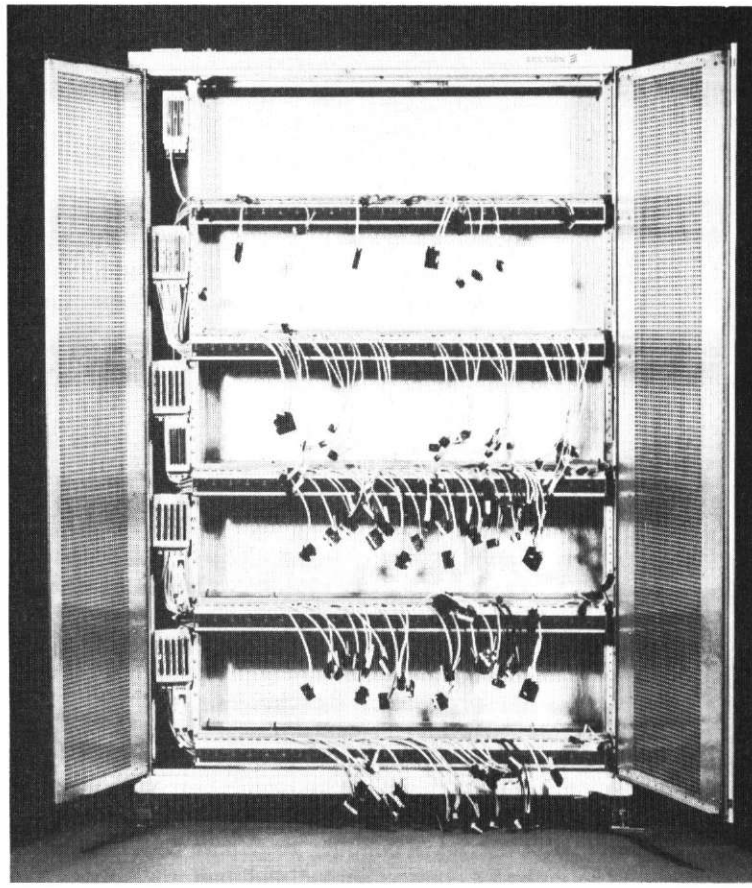

Figure 1—Factory-cabled 5-shelf cabinet ready for magazine loading

Figure 2—Delivery of 5-shelf cabinet (note minimum packaging)

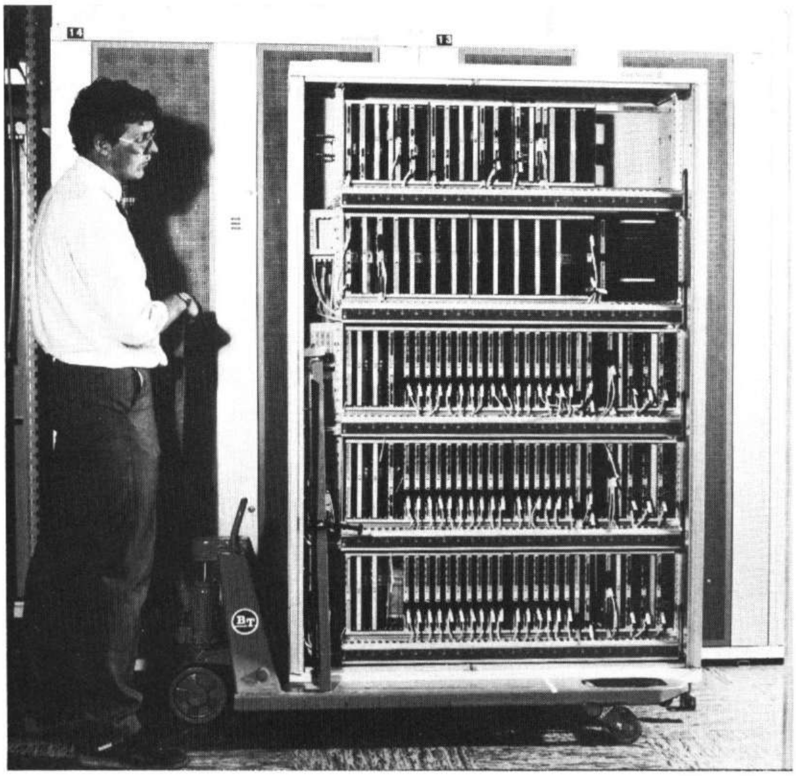

Figure 3—Five-shelf cabinet fully equipped with magazines on transport trolley

Figure 4—A pair of 5-shelf power cabinets

FIVE-SHELF UNITS

Cabinets used by Ericsson for supply and installation contracts are of 6-shelf construction. These cabinets are delivered to site empty where they are equipped with magazines, cabled and then functionally tested using the host processor.

The factory pre-testing method, however, required that cabinets must be kept upright once they had been cabled, equipped with magazines and tested, and for transportation to site. By reducing the cabinet size to 5 shelves, the total weight is reduced and centre of gravity is lowered. This enables safe handling, and the lower total height allows entry to buildings through standard height doorways. Four 100mm legs are added to the 5-shelf-cabinet to provide space underneath for inserting a pallet lifter, or flat-bed trolley, for moving the cabinets round the factory or site.

To reduce cost and lead times, a range of 'standard' cabinets has been agreed between BT and Ericsson. This range consists of 12 cabinets with fixed configurations of magazines giving a mix of analogue, digital and special equipment facilities. By suitable cabinet selection, most exchange requirements can be directly met. Fine tuning of requirements, particularly for very small exchanges, is achieved by removing magazines from a standard cabinet after factory pre-test but before delivery to site. This leaves a pre-cabled and tested shelf to which that specific magazine can be refitted if required at a later date. The factory is concerned only with building a standard range of equipment to which can be applied production line methods and to some extent a measure of pre-production. The important point to note is that cabinet construction and pre-cabling is not site specific. It is at the pre-test stage that groups of standard cabinets are assembled to match a particular exchange order and tested at EMG level.

Power cabinets are similarly factory built and pre-tested; there are three standard types each containing a 20-outlet distribution unit, a 100 Ah battery and two or three 28 A rectifiers. Power cabinets are always supplied fully equipped and can be ordered, as required, to cater for total exchange power demand. A single cabinet might be sufficient for a small exchange, but for larger exchanges several cabinets would be commoned to form a single exchange power plant. Figure 4 shows a pair of power cabinets.

RSS 256 EXTERNAL CABINET

Conventionally, remote concentrators are sited in existing telephone exchange buildings. However, to cope with situations where this is not possible, Ericsson developed the cabinet RSS, which consists of a complete exchange including MDF and power in a secure cabinet that can be sited in any convenient location and even as a roadside cabinet.

The RSS 256 external cabinets offered to BT during 1986 were developed as a 256-line ver-

sion of an earlier 128-line design which is in service in several countries. An evaluation carried out by BT Headquarters concluded that, subject to adaptive engineering to provide the SE-PRM capability (50 Hz subscribers private meter (SPM) equipment) and a battery reserve of 10 hours, a number of RSS 256 cabinets would be ordered on a trial basis. Initial deliveries of the cabinets were equipped with Ericsson 2 Mbit/s optical-fibre line terminal equipment for connection to the host exchange. At some later sites, standard BT 2 Mbit/s systems have been used for these connections.

Each RSS 256 installation consists of a pair of cabinets which are installed on a concrete base. One cabinet contains the RSS equipment magazines and MDF and the other cabinet contains rectifiers, batteries and SE-PRM (50 Hz SPM) equipment. Each cabinet is constructed of aluminium to reduce maintenance and weight, and to provide screening against radio-frequency emissions and susceptibility. This cabinet is sealed by a hinged door with an airtight sealing gasket. The internal environment is controlled by continuously running fans which circulate the air within the cabinet; the heat generated by the equipment is conducted to the outside surface by the aluminium cabinet where it is lost to the atmosphere by radiation and convection.

The cabinet is surrounded by an outer shell constructed of polyurethane foam sandwiched between glass-filled polyester resin which has good mechanical strength, to provide mechanical protection, screening from the sun's radiation and shelter from direct rainfall. The outer shell has grilles at the top and bottom which allow airflow upwards over the surface of the inner cabinet to assist in the transfer of heat to the atmosphere. To cater for extremely cold conditions, thermostatically controlled heaters may be provided in the cabinets. If desired, cabinets can be installed indoors, in which case the outer shell is replaced by a cosmetic cover.

A −50 V supply is provided for the RSS equipment by three 8 A rectifiers running off a single-phase mains supply. The rectifiers operate in load sharing mode and are dimensioned so that the full load can be supported by two rectifiers in the event of a rectifier failure. A battery reserve of 10 hours is provided by sealed lead-acid cells, and provision is made for connection of a small mobile single-phase stand-by generator to cater for longer periods of mains failure.

INSTALLATION OF FACTORY PRE-TEST EQUIPMENT

The installation techniques for both the 5-shelf pre-test equipment and the RSS 256 cabinets are broadly similar, the key stages being site preparation, delivery of the equipment, installation and finally verification of the equipment.

Site preparation is the main difference between the two applications. The 5-shelf equipment, being installed in BT exchange buildings, follows most of the normal site preparation such as clearing the necessary floor space and marking out the positions the equipment is to occupy. Because all the cables that terminate on the 5-shelf cabinet are plug ended, the site will be cabled ready to receive the equipment as part of the preparation phase; that is, all the MDF and 2 Mbit/s cables can be terminated with plugs and laid in position to connect to the cabinets on delivery. For the RSS 256, a concrete base is provided by BT onto which the cabinets are delivered. Plastic ducts are provided in the base to provide access for the cables.

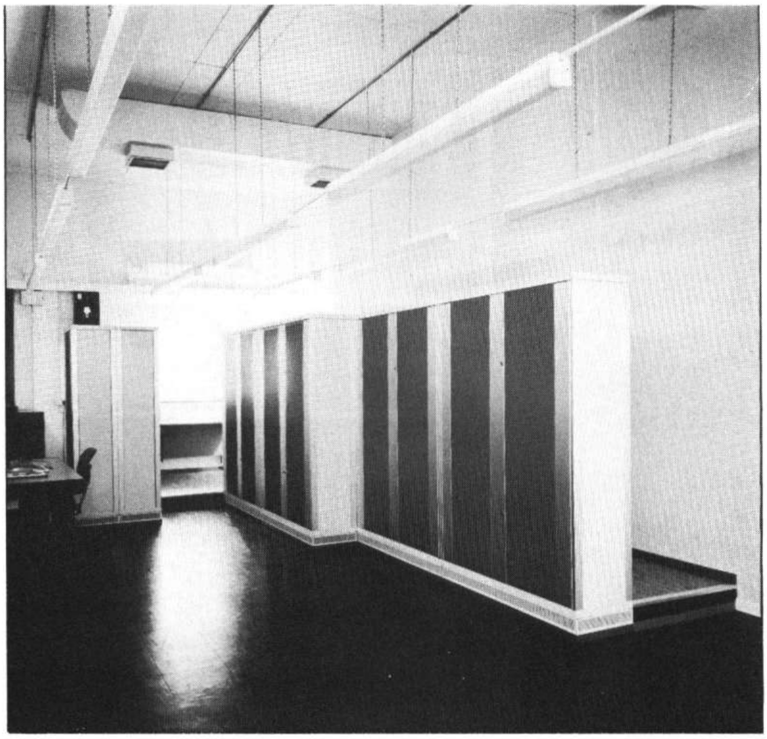

Figure 5
A 1920-line remote concentrator centre installation at Baldock

The 5-shelf cabinets and the RSS 256 are delivered to site by lorry and the supplier is responsible for locating the cabinets in their operating position. Figure 2 shows a 5-shelf delivery in progress.

After the supplier has delivered the cabinets, BT staff carry out the installation procedures. For the 5-shelf cabinets, this consists primarily of plugging up the previously prepared cables, connecting the power supply and completing the mechanical installation by fitting the final trim to the cabinets. Figure 5 shows a completed 5-shelf installation at Baldock ATE. The RSS 256 installation is completed by connecting the local distribution cables to the MDF, the optical-fibre cables to the host exchange, and the electricity supply.

For both the 5-shelf equipment and the RSS 256, this is followed by a straightforward verification procedure to ensure that no faults have arisen during transit. The installation and verification activities are of short duration and generally the unit can be brought into service within a few weeks of delivery.

APPLICATION

The 5-shelf factory pre-tested AXE10 equipment has different physical and installation characteristics, but in all other ways is standard AXE10 equipment. The 5-shelf pre-tested cabinets are equipped with standard AXE10 magazines and therefore full AXE10 features and facilities are available. The RSS 256 equipment also comprises standard AXE10 units but arranged in composite magazines for compactness. This equipment caters for up to 256 analogue lines (excluding direct dialling-in (DDI)) and provides the full range of AXE10 facilities for analogue lines; however, digital access for ISDN is not provided for the RSS 256.

The only external change to the 5-shelf cabinets is the reduced height and the legs mentioned above. Thus 5-shelf cabinets do not need rear access and can be located against a wall or placed back to back depending upon the type of accommodation available. However, the revised cabinet for the 5-shelf equipment opens up additional opportunities for applications with severe accommodation limitations. The low overall height means that 5-shelf cabinets can be installed in practically any room in any building. Access can be gained through standard height pedestrian doorways, passenger lifts can be used to reach upper floors and wall mounting capability with small 400mm depth could make even a corridor a possible installation site.

Each telephony cabinet is 1200 mm wide and 400 mm deep and, assuming a 'working' depth of 400 mm in front of the cabinet, it could be installed in a 1200 mm × 800 mm space. Each power cabinet requires even less space, being also 400 mm deep but only 720 mm wide. Both telephony and power cabinets are 1870 mm high, but some 500 mm free space above cabinets should be provided to allow adequate ventilation. The heat dissipation is around 500 W and the weight per cabinet is up to 350 kg. These latter characteristics would be the major considerations in assessing office-type accommodation applications.

The 5-shelf cabinets are raised on 100 mm legs to provide space underneath for transportation trolleys. This space can also be used for running cables between cabinets and distribution frames giving a cable floor effect at no extra cost. The use of this floor cabling technique can also reduce, considerably, the overall installation time since cables can be pre-run to marked equipment floor positions as soon as the order has been placed for the standard cabinets.

The robust construction of equipment cabinets, which is fundamental to the success of pre-test, creates another application. Provided cable runs are correctly planned and installed, then cabinets can be physically moved after installation has been completed and while live traffic is being carried. Entire suites of cabinets can be carefully raised off the floor and slowly wheeled to any desired alternative location, on the same floor, within the limitations of the external cabling. This movement is known as *hot-slide*.

Therefore, this equipment can be used for any AXE10 RSS installation irrespective of whether it is for BT installation or for supply and installation by the AXE10 contractor Ericsson. As AXE10 cabinets do not require rear access, they can be installed back to back or against a wall. Further, by employing the hot-slide technique, they can even be installed in a temporary location and then moved 'live' at a later date to a permanent location.

Factory pre-testing considerably reduces installation time, and delivery of fully cabled and equipped cabinets means that space for unpacking and assembly is not required in addition to the installation area. The 5-shelf equipment is particularly suitable for installation at sites where space is at a premium such as small rural exchanges (UAX replacement), office locations or even large buildings which might be considered 'full' using more conventional equipment practices.

The external cabinet RSS may be considered for a variety of applications, perhaps the most appropriate being to serve new residential developments; these are typically located near existing exchange boundaries and this makes the provision of service by traditional copper pair distribution from the exchange site very costly. In such cases, the provision of an RSS external cabinet may prove to be a more economic solution and will obviate any potential transmission problems. Alternatively, the RSS external cabinet can be used to replace small UAXs where savings can be made by disposing of the building and site.

EXPERIENCE TO DATE

Since the trial in 1987, in excess of 200 pre-test AXE RSSs have been ordered for installation by BT. Additionally, orders for installation by Ericsson were placed in 1988, where the short lead time was the key factor to prevent the occurrence of telephone service waiters in London. The 5-shelf RSS is now also offered by Ericsson as an option for their supply and installation orders.

The quality of the exchange deliveries has conformed to the expectations of the process and the in-service experience of the equipment conforms to that expected for AXE10.

The application of the 5-shelf RSS has been widespread throughout the BT network and BT installations have ranged from 256-line exchanges up to 20 000-line exchanges. The on-site installation and verification procedures have proved to be effective and easily undertaken by BT staff.

The first RSS 256 external cabinet installation took place at Wincle where it was used to replace a UAX 13 exchange serving a small rural

community in the hills outside Macclesfield. Delivery took place during July 1988 and the unit went into public service during August 1988. Wincle is a very exposed site, selected to give the cabinets a severe trial. Figure 6 shows the installation at Wincle. To date no service-affecting difficulties have been reported.

The RSS 256 installations following Wincle have taken place both indoors and outside and in a variety of locations. Feedback from the field has indicated that there are no major or insoluble technical or operational problems arising from deployment of RSS equipment in external cabinets in the BT network.

CONCLUSION

The objective of the AXE10 factory pre-test project was to give BT a direct labour capability for AXE10 similar to that for System X. The development of the process has met this objective, and achieved an increased quality of the delivered equipment and a significant reduction in the time between order placement and bringing the equipment into service. The pre-test process has also considerably reduced the amount of work necessary to install an AXE10 RSS. Not only is it now possible for BT to install AXE10 RSSs, but also the time and effort required to do so compares very favourably with that required to supervise an installation by the supplier. In fact, in many cases, less BT resource is actually required to do the RSS installation than supervise it.

The improved quality is mainly achieved through the factory testing which is much more comprehensive and thorough than could be done at an exchange site. The reduction in lead times is achieved in several ways. A number of activities normally done serially can be done in parallel. In particular, site preparation and MDF cabling can be done at the same time as factory production and cabling. The use of standard cabinets with standard cable configurations shortens the time to convert the BT exchange order into the required AXE10 equipment.

Considerable advances in exchange installation techniques have resulted from this process. However, there is considerable potential for the pre-test concept to evolve further towards better matching BT requirements at a reduced cost and within more closely maintained timescales. The present status of the process is the end of the beginning of its evolution.

Figure 6
RSS 256 external cabinet installed at Wincle near Macclesfield

Biographies

Andrew Papaspyru joined the then BPO in 1972. He spent his early career in TXE4 development, initially working on the type approval of the TXE4RD system and then on the TXE4A system. In 1982, he joined the evaluation team responsible for the System Y project. With the selection of AXE10 in 1985, he led one of the groups responsible for the planning and introduction of AXE10 into the network. He is now working with the intelligent network project in Network Systems Engineering and Technology. He graduated in Electrical Engineering at Imperial College, London, and subsequently obtained an M.Sc. in Solid State Physics from the same college. He is a Chartered Engineer and a Member of the Institution of Electrical Engineers.

Graham Stanley joined the British Post Office in 1958, as a Youth-in-Training, in the Bedford Telephone Area. He was successful in an open competition for Assistant Executive Engineer and moved to the Network Planning Department of Telecommunications Headquarters in 1968. During a long period working on planning and design standards for crossbar trunk switching units, he was closely involved with the introduction of the TXK4 transit switching network and the TXK1 London sector switching centres. After a few years in the Service and Performance Division (manpower planning), he joined the AXE10 Project Team in 1985. Until recently, he was the District Planning Liaison Officer for the AXE10 system with a special responsibility for developing and introducing into the network the factory pre-tested RSS equipment. He is currently working in Network Systems Engineering and Technology on networked automatic call distribution (ACD) systems and switch requirements for directory assistance modernisation.

Stuart Hawkins joined the British Post Office in 1962 as a Youth-in-Training in Bournemouth. After several years on exchange construction, he moved to Telecommunications Headquarters on promotion and was engaged on planning standards for the TXK1 system. He later moved onto System X and was part of the team responsible for the introduction of the early digital exchanges. This was followed by a short spell on ISDN specification. On promotion to level 2, he joined a small team carrying out a strategic review of the modernisation of rural group switching centres and subsequently became responsible for specification of small digital local exchange equipment including the AXE10 RSS external cabinet. He is currently a member of the digital derived services network team in Network Systems Engineering and Technology Department.

AXE10: Interworking to Analogue Exchanges

J. E. TURNER†

Although BT is rapidly introducing digital local telephone exchanges, there remains the necessity for these new exchanges to interwork with the existing analogue exchanges and with auto-manual centres. This article describes the methods adopted to cover this situation.

INTRODUCTION

British Telecom has adopted an agressive programme for the replacement of analogue local exchanges with digital versions; however, some analogue exchanges and auto-manual centres (AMCs) will remain part of the network for some years. These exchanges interwork by using a variety of analogue and pulse-code modulation (PCM) time-slot 16 (TS16) junction signalling systems and, with the exception of TXE4, there are no plans for analogue exchanges to use common-channel signalling (CCITT No. 7).

AXE10 is an exchange system designed for world-wide application but capable of adaptation to meet any telecommunication operator's specific requirements. Some development work was therefore necessary for it to support the UK analogue and PCM signalling systems. To minimise the amount of development needed, it was decided to support only the most widely used systems shown in Table 1. All systems are two-wire outgoing and incoming except where stated.

TABLE 1
Signalling Systems Supported

Analogue	Digital	Notes
Loop-disconnect 2-wire/4-wire	Loop disconnect	
DC2	DC2	
SVI/CNI leg	SVI/CNI leg	Outgoing only
SVI/CNI loop		Outgoing only
Central battery (CB)		Outgoing only
	AC8	

INTERWORKING ARCHITECTURE

The subsystem mainly concerned with inter-exchange signalling is the trunk and signalling subsystem (TSS)*, and Figure 1 shows its most important interfaces.

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* SILVERSON, R. B., and BANCROFT, J. AXE10: Architecture. *Br. Telecommun. Eng.*, Oct. 1989, **8**, p. 156

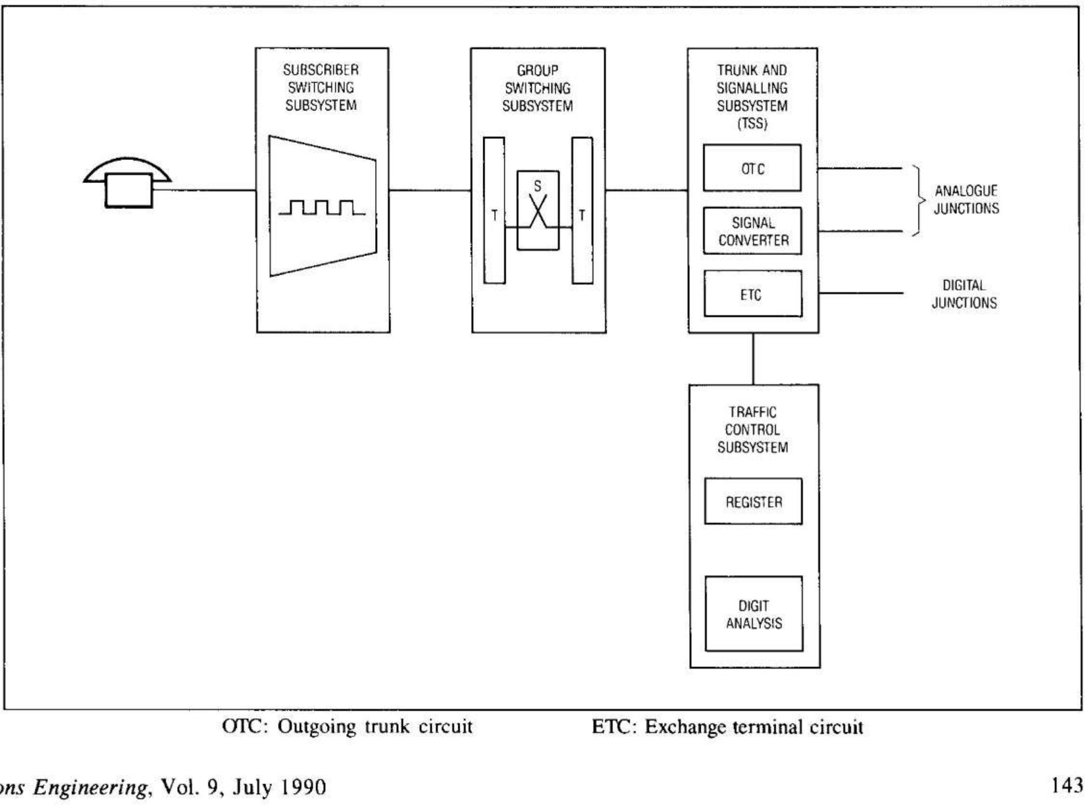


Figure 1
Most important interfaces to TSS

OTC: Outgoing trunk circuit ETC: Exchange terminal circuit

SUBSYSTEM IMPLEMENTATION

The subsystem is divided into function blocks each of which serves a particular purpose. The principal function blocks are shown in Figure 2.

Function blocks consist of a combination of hardware, central and regional processor software, although some blocks do not contain all three; for example, the bothway trunk is realised in software only. The basic function blocks are bothway trunk and outgoing trunk.

Outgoing Trunk

This function block is only employed on outgoing calls to AMCs using service interception (SVI) leg, SVI loop and central-battery (CB) analogue signalling systems (see Table 1). The outgoing trunk is instructed to select an idle circuit in the required route; a seizure signal is then sent to the distant AMC and subsequently the supervisory conditions are monitored.

Bothway Trunk

This function block can handle either incoming or outgoing traffic, the traffic directions for the circuits being stored as route data in the bothway trunk central software. For an outgoing call from the exchange, the routing digits are firstly examined and acted upon by the traffic control subsystem (TCS), the bothway trunk is then instructed to select an idle circuit in the required route. A seizure signal is accordingly sent to the distant exchange followed by digit transmission and subsequently the supervisory conditions are monitored. In the case of incoming calls, the digits received are passed to the TCS, which sets up a route through the group switching subsystem (GSS) to the required customer or, in the case of a transit call, to an appropriate outgoing junction.

HARDWARE

Figure 3 shows the hardware items used to terminate the various signalling systems supported. In each case, the GSS interface is 2·048 Mbit/s (32 × 64 kbit/s). All the items are constructed as slide-in-units which fit into a magazine.

The Ericsson equipment practice is formed of magazines mounted in cabinets. Each magazine is a self-contained unit comprising the printed wiring board assemblies, together with backplane interconnections and the power supply. The magazines are modular and constructed in a range of standard sizes to accommodate the various hardware devices.

Exchange Terminal Circuit (ETC)

A block diagram of the ETC is shown in Figure 4. This is the standard interface for digital circuits (30 speech channels with TS16 signalling). The device accepts the 2·048 Mbit/s transmission, extracts the signalling and passes

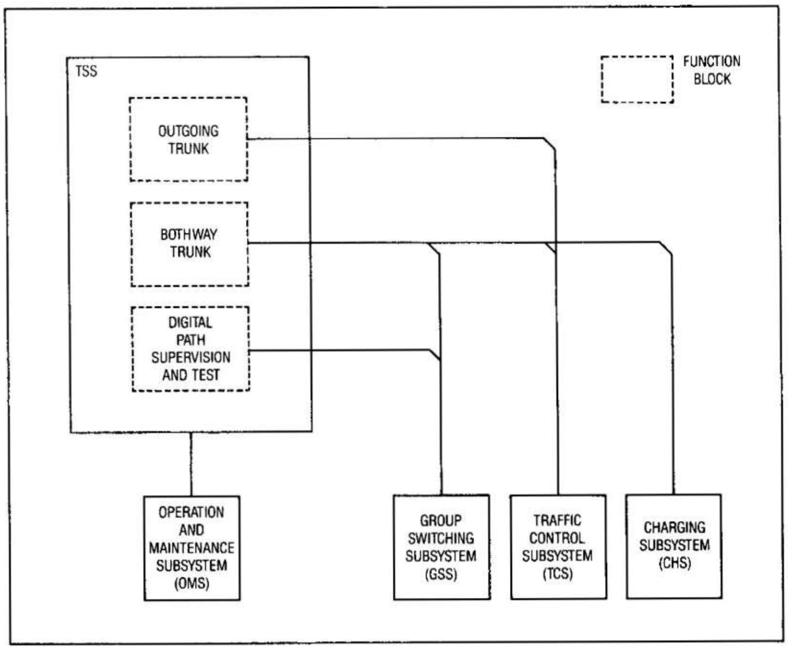
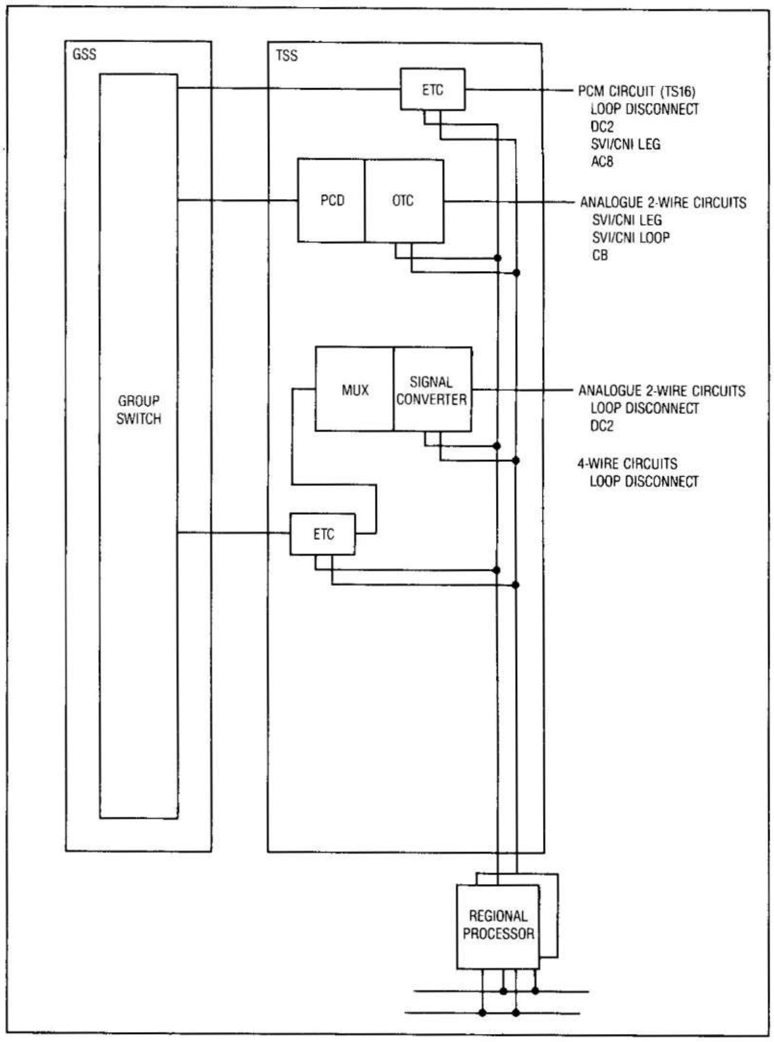


Figure 2—TSS function block and interwork diagram



GSS: Group switching subsystem
PCD: Pulse-code modulation device
OTC: Outgoing trunk circuit
TSS: Trunk and signalling subsystem
ETC: Exchange terminal circuit
MUX: Multiplexer

Figure 3—TSS hardware structure

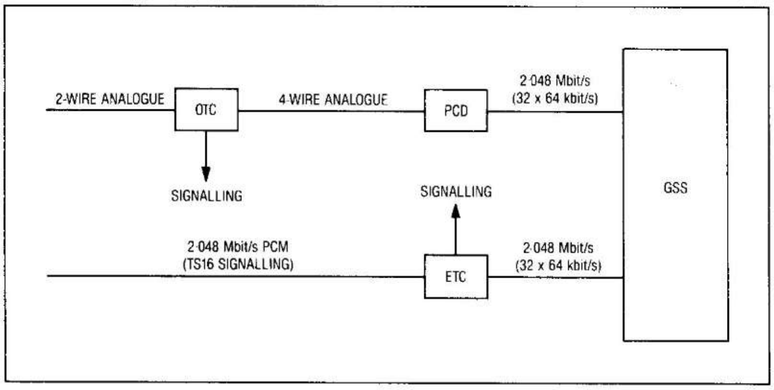


Figure 4
Exchange terminal circuit and outgoing trunk circuit/pulse-code modulation device

the speech channels to the GSS. The ETC supports loop-disconnect, DC2, SVI/CNI leg and AC8 signalling systems for both incoming and outgoing circuits, with the exception of SVI/CNI leg (outgoing only). Although the hardware is identical for all these signalling systems, they each require a different regional processor program to be associated with the ETC; hence, an ETC must be dedicated to a particular system. The only exception is that, as loop-disconnect and DC2 signalling systems use identical PCM signalling codes, they are treated as a single system by the ETC and hence may be mixed within the digital link.

The ETC is also used in association with the signal converter and multiplexer to terminate analogue loop-disconnect and DC2 circuits as described below.

Outgoing Trunk Circuit and Pulse-Code Modulation Device (OTC and PCD)

A block diagram of the OTC and PCD is shown in Figure 4. This arrangement is employed for the types of manual board analogue signalling systems which have a minority use and are likely to become obsolete within the next few years. These include SVI/CNI leg, SVI/CNI loop and CB signalling. Only outgoing circuits are supported.

Calls are routed from the GSS to the PCD where they are decoded and forwarded to the OTC over 4-wire circuits. The OTC contains a hybrid transformer which converts the transmission to 2-wire working and inserts the analogue signals. Likewise in the receive direction, analogue signals are removed by the OTC, and the PCD encodes the speech.

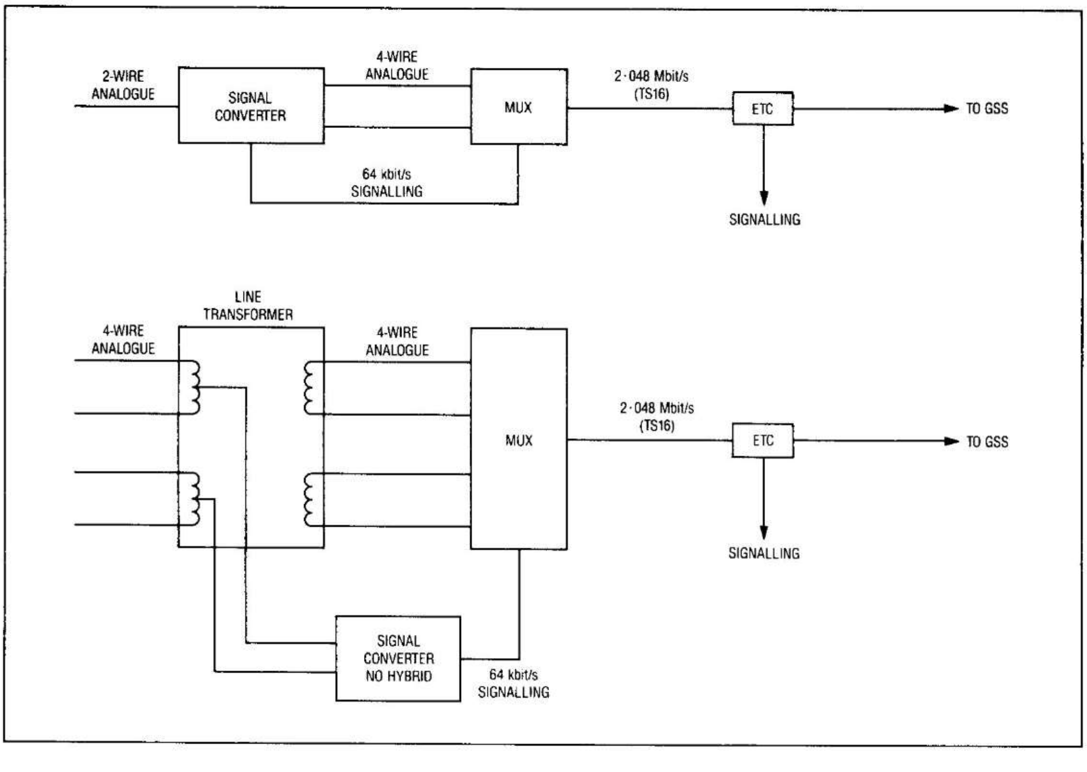
It should be noted that SVI/CNI working to any digital exchange requires the operator to adopt a slightly different procedure when extending the call in that, if the ring key is used to monitor the connection prior to extending the call, this may result in the call being released. To prevent this, service-interception manual-board relay-sets (for example, AT4006) have been modified to inhibit the signal to line.

Signal Converter and Multiplexer

A block diagram of the signal converter, which is a newly developed device, and multiplexer (MUX) is shown in Figure 5. Signal converters are used for loop-disconnect and DC2 signalling systems. Both incoming and outgoing circuits are supported.

Calls are routed from the GSS to the ETC, the signalling is inserted and they are then forwarded to the MUX in standard 2·048 Mbit/s PCM format (signalling in TS16). Here the 64 kbit/s signalling is extracted, and the speech is decoded and passed over a 4-wire circuit to the signal converter. A hybrid transformer converts the transmission to 2-wire

Figure 5
Signal converter and multiplexer



working, and the signalling from the MUX is converted to analogue and inserted in the 2-wire signal. Likewise, in the incoming direction, analogue signals are removed by the signal converter and the MUX encodes the speech.

For 4-wire working, the junction is terminated on a line transformer. This extracts the analogue signalling from the phantoms to a signal converter (minus hybrid), where it is converted into 64 kbit/s and injected into the MUX, as in the case for 2-wire working.

The advantage of using the signal converter/multiplexer rather than an OTC/PCD type arrangement is when it is required to upgrade the analogue circuits to PCM. No new hardware is needed; the signal converter/multiplexer is recovered for reuse and the junction connected direct to the ETC.

FLEXIBILITY AND DIMENSIONING

Exchange Terminal Circuit

A magazine can hold up to 16 ETCs which need not be all of the same signalling type. However, all the circuits within each ETC must use the same signalling system, but, as mentioned above, loop-disconnect and DC2 systems are considered to be identical.

Outgoing Trunk Circuit and Pulse-Code Modulation Device

An OTC magazine handles 16 circuits which must be all of the same type; that is, SVI/CNI leg, SVI/CNI loop or CB. The 16-circuit group is then connected to a PCD magazine which accepts the output from two OTCs. These two circuit groups can be of different types.

Signal Converter and Multiplexer

A signal converter magazine is installed in a double shelf and can accommodate a total of 30 circuits, consisting of 15 printed-circuit boards with two circuits per board. Separate boards are used for loop-disconnect or DC2 signalling systems and for incoming and outgoing circuits. However, all types of board can be mixed within a magazine thus giving complete flexibility. The magazine can also, when required, include DC/DC converters to provide the additional voltages needed for these analogue signalling systems; for example, −100 V manual-hold loop-disconnect and +50 V operator override DC2.

The signal converter is connected to a MUX which, therefore, is also capable of handling any combination of loop-disconnect, DC2, incoming or outgoing circuits.

Actions Under Fault Conditions

Faulty transmission when using digital systems can result in poor speech quality and incorrect signalling. In addition, invalid TS16 codes may be received on occasions. All calls are monitored by function blocks in TSS and, in these circumstances, an appropriate action is taken; for example, busy device, release connection, run time-out or indicate a fault.

TRAFFIC TYPES

Circuits can be used for different types of traffic and in AXE10 the following types are programmable.

Type I	Basic call
Type IB	Satellite type manual hold and coin-and-fee-check signals
Type IC	Plug-controlled supervision manual hold and coin-and-fee-check signals
Type ID	Code only calls with trunk coin-and-fee-check signals
Type IE	Code only calls with coin-and-fee-check signals (no manual hold)
Type IIC	Metering-over-junction calls
Type III	Calls transferring operator signals (that is, operator override and howler)

SUMMARY

The arrangements adopted enable AXE10 to support the majority of analogue signalling systems currently in use. For the most commonly used systems, a new development was undertaken which should minimise rearrangements when conversion to TS16 signalling is necessary. It also gives considerable flexibility in the use of loop-disconnect and DC2 signalling systems for both incoming and outgoing circuits.

ACKNOWLEDGEMENTS

The author wishes to thank his colleagues in the AXE10 Project Section and within Ericsson Ltd. for the provision of information contained in this article.

Biography

John Turner joined the British Post Office in 1950 as a Youth-in-Training in the Telecommunications Headquarters Circuit Laboratory. After National Service, he returned to work in the development of subscriber trunk dialling equipment. In 1959, he was successful in the limited competition for Assistant Executive Engineer and was engaged on relay design. This was followed by a move to tester design for PABXs and the transit network. Promoted to Executive Engineer in 1969, he moved to data systems planning and then returned to telephony to work on digital exchange facilities. In 1982, he joined the initial evaluation team responsible for the System Y project and has remained with it. He has a City and Guilds of London Institute Full Technological Certificate and is a Member of the Institution of Electronics and Electrical Incorporated Engineers.

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RULE CHANGES

The proposed amendments to the Rules announced in the April 1990 issue of the *Journal* have been suspended until the organisational changes under Project Sovereign have been established. Council is currently operating under the arrangement whereby current Council Members are serving a minimum of one further year of office.

IBTE CONGRESS

The IBTE Congress held on 30 and 31 May 1990 at the Chartered Insurers Institute in London was very successful. The Congress laid the groundwork for the renaissance of the IBTE and should prove to be of great benefit to the current and future generations of Members, as well as BT. A detailed report of the Congress will be published later.

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Contents

VOL 9 PART 2 JULY 1990